

TAPR HF Special Interest Group Archive, October 1994 through December 1995

October 1994 through December 1995

TAPR's forum for experimentation and development of HF digital communications, including modulation, coding, protocols, DSP platforms, modems, propagation, and on-air testing.

This volume contains 15 original monthly files and 777 messages, arranged chronologically. The PDF is a readable access copy: MIME transfer encodings are decoded, HTML bodies are converted to plain text, and non-text attachments are represented by descriptive notes. The raw source text is supplied separately with the Internet Archive item.

Month	Messages	Original source file
1994-10	59	hfsig.9410
1994-11	44	hfsig.9411
1994-12	57	hfsig.9412
1995-01	63	hfsig.9501
1995-02	58	hfsig.9502
1995-03	56	hfsig.9503
1995-04	48	hfsig.9504
1995-05	1	hfsig.9505
1995-06	40	hfsig.9506
1995-07	67	hfsig.9507
1995-08	65	hfsig.9508
1995-09	70	hfsig.9509
1995-10	68	hfsig.9510
1995-11	18	hfsig.9511
1995-12	63	hfsig.9512

October 1994

Original source file: hfsig.9410

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1994-10/TAPR-HF_1994-10.txt

1. HF Digital Bibliography

From: "Rick Whiting" <Rick_Whiting@ATK.COM>
Date: 10 Oct 1994 14:51:40 -0600
To: "Post hfsig TAPR" <hfsig@tapr.org>
Cc: Bob_Hunter@gateway1.ATK.COM, Darrell_Sawyer@gateway1.ATK.COM
Message-ID: <9410101943.AA06809@ATK.COM>
Rendered body type: text/plain

Subject: Time:2:33 PM

OFFICE MEMO HF Digital Bibliography Date:10/10/94
Someone on the Ham Digital newsgroup was looking for info on PACTOR the other day. This prompted me to go through my files and prepare a bibliography. This was limited, for the most part, to the ham periodical literature. I specifically excluded professional journals and books as these can be easily located through normal library research techniques. Thought it might be interesting to the hfsig too. Did I miss any "good ones"?

Amateur Radio Periodical Literature Bibliography

HF DIGITAL COMMUNICATIONS

W0TN, Rev. 0, 10-07-94

ALE (Automatic Link Establishment)

Adair, Robert T., KA0CKS, David Peach, and Dennis Bodson, W4PWF, "The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)," Part 1, The National Communications System, The Federal Standards Development Process, and the Basic Definition of Federal Standards 1045 through 1054, QEX, July 1993, pp. 3-8.

Wortendyke, David, N0WGC, Chris Riddle, KB0HNM, and Dennis Bodson, W4PWF, "The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)," Part 2, Compact Disc for Testing HF ALE Radios, QEX, Aug. 1993, pp. 9-14;

Brewster, Larry, N0RNX, and Dennis Bodson, W4PWF, "The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)," Part 3, Where are the Federal Standards for HF ALE Radio Networking Going?, QEX, Sept. 1993, pp. 14-19;

Sutherland, David, and Dennis Bodson, W4PWF, "The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)," Part 4, Network Simulation for the Radio Amateur, QEX, Oct. 1993, pp. 13-18.

Smith, Paul C., K3ZMO, and Dennis Bodson, W4PWF, "The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)," Part 5, An Amateur's Practical Approach to HF ALE Radio Systems, QEX, Nov. 1993, pp. 9-12.

Redding, Christopher, and Dennis Bodson, W4PWF, "The Growing Family of Federal Standards for HF Radio Automatic Link Establishment (ALE)," Part 6, Federal Standard 1049--The Future of ALE Operation in Stressed Environments, QEX, Dec. 1993, pp. 7-9.

Levreault, Bob, W1MM, and Ken Wickwire, KB1JY, "Some Recent Amateur Use of Federal Standard Automatic Link Establishment (ALE) Signaling," Proceedings of the 11th ARRL Conference on Digital Communications, Nov. 1992, pp. 32-39.

AMTOR/SITOR

Bingemer, Armin, DK5FH, "AMC--The AMTOR Controller," QEX, Feb. 1992, pp. 3-11.

Henry, Bill, K9GWT, "New AMTOR Mode," CQ, Nov. 1989, pp. 36-40.

Huslig, Michael J., "Unique Identification Signals For CCIR 625," Proceedings of the 12th ARRL Conference on Digital Communications, Sept. 1993, pp. 28-29.

Martinez, Peter, G3PLX, "AMTOR, An Improved Error-Free RTTY System," QST,

June 1991, pp. 25-27.

Newland, Paul, AD7I, "An Introduction to AMTOR," QST, July 1983, pp. 11-13.

Newland, Paul, AD7I, "Algorithms and Methods For SITOR/AMTOR Systems," QEX, July 1988, pp. 9-12.

Young, Teresa, Stephen Rieman, and David Wortendyke, NØWGC, "A Preview of HF Packet Radio Modem Protocol Performance," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 152-155.

"HF Modem Over-The-Air Performance Test Report, Globe Link Corporation HF Modem Model GLC-1000A," Fed. Emergency Mgmt. Agency (FEMA), Washington, DC, 20472, Feb. 22, 1993.

"HF Modem Over-The-Air Performance Test Report, Ascom Radiocom Ltd. ARTOR HF Modem," Fed. Emergency Mgmt. Agency (FEMA), Washington, DC, 20472, April 12, 1993.

APLINK (AMTOR to AX.25 Packet Link)

Mortensen, Jim, N2HOS, "APLINK: The Delivery System--Part 1," QST, Oct. 1991, p.73.

Mortensen, Jim, N2HOS, "APLINK: The Delivery System--Part 2," QST, Nov. 1991, pp. 81-82.

CLOVER

Horzempa, Stan, W1LOU, "CLOVER: The Future of HF Data Communications?," QST, Jan. 1993, p. 107.

Petit, Raymond C., W7GHM, "The 'Cloverleaf' Performance-Oriented HF Data Communication System," QEX, July 1990, pp. 9-14.

Petit, Raymond C., W7GHM, "The CLOVER-II Communications Protocol Technical Overview," Proceedings of the 11th ARRL Conference on Digital Communications, Nov. 1992, pp. 47-51.

Petit, Raymond C., W7GHM, "CLOVER-II: A Technical Overview," Proceedings of the 10th ARRL Conference on Digital Communications, Sept. 1991, pp. 125-129.

Westhuizen, Mike Van Der, ZS6UP, "A Practical Comparison Between Clover and Pactor Data Transfer Rates," CQ, Vol. 50, No. 2, Feb. 1994, pp. 40, 42.

Young, Teresa, Stephen Rieman, and David Wortendyke, NØWGC, "A Preview of HF Packet Radio Modem Protocol Performance," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 152-155.

"HF Modem Over-The-Air Performance Test Report, HAL Communications Corp. PCI-4000 HF Modem," Fed. Emergency Mgmt. Agency (FEMA), Washington, DC, 20472, July 6, 1993.

G-TOR

Anderson, Phil, WØXI, Michael Huslig, Glenn Prescott, WBØSKX, and Karl Metcalf, WK5M, "G-TOR The New, Faster HF Digital Mode for the KAM-Plus," Kantronics, Inc., 1202 E. 23rd St., Lawrence, KS 66046, U.S.A., 1994.

Huslig, Mike, Phil Anderson, Karl Metcalf, and Glenn Prescott, "G-TOR: The Protocol," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 49-79.

Huslig, Richard, and Phil Anderson, "GMON--A G-TOR Monitoring Program for PC Compatibles," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 80-85.

Prescott, Glen E., WBØSKX, and Phil Anderson, WØXI, "A Theoretical Evaluation of the G-TOR Hybrid ARQ Protocol," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 86-89.

Prescott, Glenn, WBØSKX, Phil Anderson, WØXI, Mike Huslig, and Karl Metcalf, WK5M, "G-TOR: A Hybrid ARQ Protocol for Narrow Bandwidth HF Data Communication," QEX, May 1994, pp. 12-19.

Prescott, Glenn E., WBØSKX, and Phil Anderson, WØXI, "Hybrid ARQ For HF Data

Transmission," Communications Quarterly, Vol. 4, No. 3 (Summer), 1994, pp. 31-40.

Packet (AX.25) [Not recommended for HF]

AX.25 Amateur Packet-Radio Link-Layer Protocol, Version 2.0, Oct. 1984, American Radio Relay League, 225 Main St., Newington, CT 06111, U.S.A.

Karn, Phil, KA9Q, "Spectral Efficiency Considerations for Packet Radio," Proceedings of the 10th ARRL Conference on Digital Communications, Sept. 1991, pp. 62-66.

McLarnon, Barry D., VE3JP, "HF Packet: Where Do We Go From Here?," Proceedings of the 6th ARRL Conference on Digital Communications, Aug. 1987, pp. 126-133.

Young, Teresa, Stephen Rieman, and David Wortendyke, N0WGC, "A Preview of HF Packet Radio Modem Protocol Performance," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 152-155.

PACTOR

Clas, Martin, DL1ZAM, and Peter Mack, DL3FCJ, "PTC--The PACTOR Controller," QEX, Oct. 1991, pp. 7-11.

Helfret, Hans-Peter, DL6MAA, and Ulrich Strate, KF4KV, "PACTOR--Radioteletype With Memory ARQ and Data Compression," QEX, Oct. 1991, pp. 3-6.

Reedy, Gwyn, W1BEL, "PACTOR: An Overview of a New and Effective HF Data Communication Protocol," Proceedings of the 11th ARRL Conference on Digital Communications, Nov. 1992, pp. 62-64.

Westhuizen, Mike Van Der, ZS6UP, "A Practical Comparison Between Clover and Pactor Data Transfer Rates," CQ, Vol. 50, No. 2, Feb. 1994, pp. 40, 42.

Young, Teresa, Stephen Rieman, and David Wortendyke, N0WGC, "A Preview of HF Packet Radio Modem Protocol Performance," Proceedings of the 13th ARRL Conference on Digital Communications, Aug. 1994, pp. 152-155.

"HF Modem Over-The-Air Performance Test Report, Globe Link Corporation HF Modem Model GLC-1000A," Fed. Emergency Mgmt. Agency (FEMA), Washington, DC, 20472.

PACTOR II

Rink, Dr. Tom, DL2FAK, "Update on the Development of PacTOR-2," Special Communications Systems, Rontgenstr. 36, 63454 Hanau, Germany.

MISCELLANEOUS

Baiya, Harun, David Balson, and Gary Garriott, "Digital Radio Technology and Applications," Proceedings of Intl. Workshop organized by the Intl. Dev. Res. Center, Volunteers In Technical Assistance (VITA), and UN University, Nairobi, Kenya, Aug., 1992.

Cadmus, Clayton, KA0GKC, and Bruce L. Meyer, W0HZR, "FSK Signal Monitor," Communications Quarterly, Vol. 3, No. 2 (Spring), 1993, pp. 45-50.

Chandler, Dr. Alan, K6RFK, "HF Modems for Data Transmission, A look at the Differences Between Commonly Used Designs," Communications Quarterly, Vol. 1, No. 2 (Spring) 1991, pp. 74-78.

Clark, Tom, W3IWI, "Comments on HF Digital Communications Part 1--Link Level Issues," Proceedings of the 9th ARRL Conference on Digital Communications, Sept. 1990, pp. 265-270.

Clark, Tom, W3IWI, "Comments on HF Digital Communications Part 2--Data Protocol Issues," Proceedings of the 9th ARRL Conference on Digital Communications, Sept. 1990, pp. 271-274.

Henry, Bill, K9GWT, and Ray Petit, "HF Radio Data Communication, CW to CLOVER," Communications Quarterly, Vol. 2, No. 2 (Spring), 1992, pp. 11-24.

Johnson, Lyle V., WA7GXD, "Thoughts on an Adaptive Link Level Protocol," Proceedings of the 8th ARRL Conference on Digital Communications, Oct. 1989,

pp. 97-107.

Kinsner, W., VE4WK, "Lossless Data Compression Algorithms for Packet Radio," Proceedings of the 10th ARRL Conference on Digital Communications, Sept. 1991, pp. 67-75.

Karn, Phil, KA9Q, "Toward New Link-Layer Protocols," QEX, June 1994, pp. 3-10.

McIntyre, Lewis F., "Error Correction In Data Transmission," Communications Quarterly, Nov. 1990, pp. 88-94.

Nagle, John J., K4KJ, "Diversity Reception: An Answer to High Frequency Signal Fading," Ham Radio, Nov. 1979, pp. 48-55.

Shapiro, Garry, NI6T, "Optimizing the PK-232MBX for RTTY and AMTOR," Communications Quarterly, Vol. 3, No. 1 (Winter), 1993, pp. 83-91.
Sykes, B., G2HCG, "The Enhancement of HF Signals By Polarization Control," Communications Quarterly, Nov. 1990 pp. 23-28.

Wickwire, Ken, KB1JY, "The Status and Future of High Frequency Digital Communications," Part 1: Overview, QEX, June 1992, pp.3-14.

Wickwire, Ken, KB1JY, "The Status and Future of High Frequency Digital Communications," Part 2: HF Modems and Their Performance, QEX, July 1992, pp. 3-15.

Wickwire, Ken, KB1JY, "The Status and Future of High Frequency Digital Communications," Part 3: Simulating The Performance Of HF Digital Networks, QEX August 1992, pp. 12-17.

Wickwire, Ken, KB1JY, "The Status and Future of High Frequency Digital Communications," Part 4: Where Is HF Digital Networking and Where Is It Going, QEX, Oct.1992, pp. 10-18.

"Low-Cost Digital and Packet Radio Techniques: Operation and Applications," Volunteers in Technical Assistance (VITA), Arlington, VA, Spring 1994.

Proceedings of the 3rd FEMA MF/HF Packet Radio Conference, Fed. Emergency Management Agency (FEMA), Washington, D.C. 20472, April 1992.

2. High Speed HF modem progress

From: "Johan Forrer
Date: Mon, 17 Oct 1994 09:07:54 PST8PDT
To: HFSIG@tapr.org
Message-ID: <54FDDFE3172@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

I have put a posting out on Usenet for more on an "HF channel simulator" and have received some (not much though) feedback. There apparently are narrow band and wide band simulators, the latter being for spread spectrum (SS) applications and there are some interest presently by the military in this. We are mostly interested in a narrow-band simulator, that, by the way, can be a very complex thing if indeed you want to simulate the behavior at a particular location at a particular instant in time. Those of you that have played with "IONCAP" and similar program will have some idea what this involves. There is also a much simpler simulation using the CCIR "good" and "poor" recommendations. These models allows one to adjust S/N settings and allows the modem to be evaluated in a little as 0.1 dB steps. I have some scanty specs on the CCIR model can anyone help with obtaining a copy of this? (It is in the Dubrovnic plenary session, ca. 1989).

Otherwise, I am in the process of bringing up a 100 baud QPSK single tone modem. It is possible that such an arrangement using two parallel channels will allow robust 400 bps using a regular RTTY channel bandwidth (without coding with Golay (24,12) the rate would be 200 bps). This code will run on the Cardinal soundcard. The CCIR channel simulator will also run on the Cardinal sound card.

Does anyone have any ideas on spread spectrum implementations for HF?

I have not heard much from anyone being interested in participating in code development. You are missing out on some very interesting and educational opportunities. If you are interested, let us hear from you.

73's

Johan
KC7WW

3. BBS with 2 HF ports?

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Mon, 17 Oct 1994 14:53:23 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410171953.0AA21349@allanon.ia.net>
Rendered body type: text/plain

Hi

I realize this is not the current topic, but a list of BBSs which have ports on 2 or more frequencies would be of some value. The BBSSIG is making a table of states/substate identifiers and a list of these hf net interconnects would be of value.

Does someone have such a list?
Could we build one for first hand information?

I am willing to compile this list if that will help.

73 Barry

4. Re: [HFSIG:14] High Speed HF modem progress

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Tue, 18 Oct 1994 10:13:11 -0700 (PDT)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941018095228.29382K-100000@daffyduck>
Rendered body type: text/plain

Johan,

Forgive the delay in responding to your recent postings. They contain a lot of information and it's taken me some time to hoist in even part of it!

>

> I have put a posting out on Usenet for more on an "HF channel simulator" and
> have received some (not much though) feedback. There apparently are narrow

I did a literature search on the University of Washington's online catalog and couldn't find a single reference to HF channel simulators. I haven't had a chance to manually search the IEEE index.

> allows the modem to be evaluated in alittle as 0.1 dB steps. I have some
> scanty specs on the CCIR model can anyone help with obtaining a copy of
> this? (It is in the Dubrovnic plenary session, ca. 1989).

If you can send some more information I'll try the UoW engineering library.

>

> Otherwise, I am in the process of bringing up a 100 baud QPSK single tone
> modem. It is possible that such an arrangement using two parallel channels
> will allow robust 400 bps using a regular RTTY channel bandwidth (without
> coding with Golay (24,12) the rate would be 200 bps). This code will run
> on the Cardinal soundcard. The CCIR channel simulator will also run on the
> Cardinal sound card.

The sound card seems like a good approach. TAPR's DSP-93 is pricey and has a long delivery time.

>

> Does anyone have any ideas on spread spectrum implementations for HF?
>

I've done some work with narrow-band spread spectrum transmission of digital video over wireline. (This was some years ago and so we were doing everything in hardware. These days a run of the mill DSP could do the same thing.) This approach may be applicable to HF but I have no experience with this application. One observation I could make is that the scheme we used seemed to be quite sensitive to the phase distortion of the channel. I suspect the phase distortion of the HF channel is going to present a challenge regardless of the modulation scheme we pick.

> I have not heard much from anyone being interested in participating in code
> development. You are missing out on some very interesting and educational

> opportunities. If I you are interested, let us hear from you.

Please count me in! What's the first task you have in mind?

Hugh

5. HF Channel Simulator

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Tue, 18 Oct 94 13:35:00 EDT
To: hfsig <hfsig@tapr.org>
Message-ID: <2EA40772@arrl.org>
Rendered body type: text/plain

Johan -- I'm interested in the HF channel simulator project. I've ordered a copy of CCIR Rec. 520-1, which I think includes specifications for CCIR standard channels. In Kantronics' May, 1994 QEX article about G-TOR, they refer to an HF channel simulator they used to test their G-TOR system. This simulator used a TMC320C30 and implemented the "good, moderate, poor and flutter fading channels prescribed by the CCIR as recommended for simulator test channels." That seems like a good place to start. I should have the document in a week or so and will report here on what I find out about the CCIR standards.

Then again, maybe I'm behind the power curve and the rest of you already know all about these CCIR recommended channels?

-- Jon, KE3Z

6. RE: [HFSIG:17] HF Channel Simulator

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Tue, 18 Oct 94 13:43:00 EDT
To: hfsig <hfsig@tapr.org>
Message-ID: <2EA40A15@arrl.org>
Rendered body type: text/plain

Oh, I should also mention that the November QEX editorial is about the need for HF channel simulation and refers to this mailing list, with instructions on how to access it. Maybe that'll drag a few more people into the mix.

-- Jon

7. Re: [HFSIG:16] Re: High Speed HF modem progress

From: "Johan Forrer
Date: Tue, 18 Oct 1994 11:49:42 PST8PDT
To: hfsig@tapr.org
Message-ID: <56A91835976@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Hugh,

I must apologize that we jumped into the deep end - if you are somewhat overwhelmed by the details, hang in there.

The channel simulator would play an important and necessary role during evaluation. From re-reading Ken Wickwire's excellent QEX articles (ca. 1992) and also from Freeman's handbook, I found that CCIR rep. 549-2 (XVI th Plenary Assembly, Dubnovrik, 1986, Vol III) contains a wealth on this topic. Likewise, have found that our library does not have these reports, so hopefully, someone else may be able to help out. I note that Jon may be able to help there - we will appreciate any efforts to locate this source of information.

It is obvious that a full-blown channel simulator is a very complex thing, perhaps a project that someone could pursue, however, a simpler model, that perhaps includes multipath with variable delay, fading rate, channel bandwidth and S/N, may suffice to get us on the right track. It would be really great if we could establish a few norms for the known modulation schemes under such conditions and then direct efforts into those that are most promising.

Regarding the DSP platform - the TAPR DSP-93 is a fine design and in time will establish a good base of users and applications. Personally I have found the DSP sound card approach quite attractive for development work and have collected a number of useful tools along the way. I also am interested in the possibilities of integrating this hardware into the mainstream OS. You probably are aware that there are some very innovating DSP- related API components under development that will come on-line in the near future.

>From your comments on SS, it appears that I need to do more homework on SS. It appears that once we get it to work, it may be the solution to many of our present problems. One question - does it mean that if we have found our efficient modulation scheme, that it is just a matter of implementing it on top of the SS coder/decoder? (pardon my ignorance).

Johan
KC7WW

73/Rick WOTN

Barry, have you gotten feedback yet on your posting on "2-port" HF capability?

Take care,

Johan
KC7WW

10. H.F modems

From: Charles Brain <g4guo@dircon.co.uk>
Date: Wed, 19 Oct 1994 21:32:15 +100 (BST)
To: hfsig@tapr.org
Message-ID: <Pine.3.89.9410192101.A23693-0100000@tdc.dircon.co.uk>
Rendered body type: text/plain

Hello,

A good starting place for H.F modems is Proposed Federal Standard 1052. The main body of the standard describes a serial tone modem, Appendix A contains details of a parallel tone modem and Appendix B the Data Link Protocol.

Also anyone developing Mil-Std-188-141A/ FS1045/1046 can get a DOS/Windows package to test it using a sound blaster card from ntia.its.bldrdoc.gov in dist/ale-cd.

My main concern is on what frequencies these modems will be used? They use voice bandwidths so the C.W/RTTY frequencies are probably out and if you go into the voice portions you will be accused of being a slow scan operator!

I would like to see channel adaption discussed, why bother getting the last db of performance when a lot more can be gained by moving to a new channel!

I would be more interested in very robust slow speed modems i.e chirp. The company I work for have developed a chirp modem but I believe it uses about 5x56000 dsps.

Although I have ordered a DSP93 board I tend to think it really isn't powerful enough for more advanced modems.

The 'in' thing appears to be parallel modems using soft decoding and Trellises.

Regards Charles

```
-----
| 73 Charles H. Brain G4GUO          |
| E-mail g4guo@dircon.co.uk        |
| Tel Day (0245)353221 Xtn 3254    |
| Eve (0245)469382                  |
| TCP/IP g4guo.ampr.org AX25 g4guo@gb7esx |
| Snail 2 Daisy Court, North Springfield, Chelmsford. |
| ESSEX. CM1 5QU. U.K              |
|-----
```

11. found CCIR 549-2

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Wed, 19 Oct 1994 15:51:14 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410192051.PAA09357@allanon.ia.net>
Rendered body type: text/plain

Hi all,

I just received a copy of CCIR Report 549-2. It was faxed in here and is readable, but would not survive another fax. A paper copy is on the way and should be here in a few days.

Quoting the Report

"This Report describes a Gaussian-scatter HF channel model for multiplicative distortion, the experimental method used to confirm the validity and the bandwidth limitation of this model, and one implementation of an HF simulator based on this model."

The entire Report is about 9 pages and contains 3 figures, 2 nasty looking equations, 20 references, and 2 Annex(s) with 2 more figures.

Figure

1. Power Spectra of the multipath components of a CW signal.
2. Block diagram of HF ionispheric channel models.
3. Tap-gain spectra in validated Gaussian-scatter model.
4. Recording the channel characteristics.

5. Simulation of the channel using recorded characteristics.

Johann, I'll mail you a copy of the Report when I get the real copy.

FREE COPIES!

We'll sort of. The first 5 people who request a copy will get one free.
Please be able to make some use of it!

Hope this helps,

Barry wa0rjt

barry@ia.net

sys_bjb%afds@hobbes.cca.rockwell.com

I read this in the evenings.

Stared at continuously 8-5
Central time!

12. Re: [HFSIG:22] H.F modems

From: "Johan Forrer
Date: Wed, 19 Oct 1994 14:36:26 PST8PDT
To: hfsig@tapr.org
Message-ID: <58559841185@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hello Charles,

Nice to hear from you. You certainly are bringing up a number of issues that have resulted in the formation of this HFSIG - good for you!

Let us entertain a few notions and bring everyone up to speed: The HF modems that Charles are mentioning are all in the family of standards built around MIL-STD-188. This includes a 39 parallel-tone, 16-parallel tone, and 8-ary FSK. The latter one is also known as the ALE format. NTIA makes a CD-ROM test source available, primarily for testing ALE equipment. Some of these modems are capable of 2400 bps (single channel or time-divison multiplexed) and are not shy about fully utilizing a 3kHz bandwidth channel. Several outfits are producing such modems, and they are quite pricey - at least out of reach for the usual radio amateur. In my opinion - (for what its worth) this is more or less brute force bandwidth & power to meet a specific objective. It is my belief that it is our challenge as radio amareurs to come up with something a bit more refined, superior in performance, and affordable. However, these documents makes a good source of information.

The chirp modulation modems that Charles mentions, implements a form of narrow-band SS. I have a couple of papers from the IEE on the theory and implementation details (let me know if you are interested in the references). If you have ever dealt with chirp and chirp-z transforms, you will have some idea what this is all about. This kind of innovation gives me hope that something as innovative will eventually lead to better solutions.

You are quite correct about the modern trend toward parallel modems and clever coding, and we are heading that way too - though, hopefully, by doing our homework first to find out what basic modulation scheme to use - this is the whole idea behind the simulator. Hopefully we will have this basic knowledge shortly and then be able to move on to working on coding theory and eventually protocol.

In an earlier posting I provided a guestimate of the capabilities of a few modulation schemes (some including coding) and shown that a typical single-DSP platform would perhaps (just a guess) be able to handle a x4 parallel situation (using fairly conventional modulation schemes). I hope that we would be able to do at least as good as the CLOVER II system. I am pleased to hear that you took the plunge with the DSP-93 - keep us posted on your developments.

Thanks for your input - keep them coming.

73's

Johan

13. Re: [HFSIG:23] found CCIR 549-2

From: "Johan Forrer
Date: Wed, 19 Oct 1994 14:38:55 PST8PDT
To: hfsig@tapr.org
Message-ID: <585641372D4@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Barry,

That is great news - looking forward seeing the document.

Johan

14. Re: [HFSIG:25] Re: found CCIR 549-2

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Wed, 19 Oct 1994 18:05:16 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410192305.SAA10200@allanon.ia.net>
Rendered body type: text/plain

Johan

Sorry I missed on your name!

Send me a good address.

Do you know anyone that works for Rockwell in Dallas? I got the company librarian here in Cedar Rapids to make a few calls. Seems like there might be other items of interest in the lib there, but searching via long distance is not effective. They are very good at the library tasks but I hate to take up their time. We get regular ethics lectures...

I am also aware that the comm division does ALE and other hf "special" tasks. They might be able to help but are usually reluctant to go public with their knowledge.

73 Barry wa0rjt

15. Re: [HFSIG:19] Re: High Speed HF modem progress

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Wed, 19 Oct 1994 16:36:00 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941019161433.29382f-100000@daffyduck>
Rendered body type: text/plain

On Tue, 18 Oct 1994, Johan Forrer FL wrote:

>
> our present problems. One question - does it mean that if we have found our
> efficient modulation scheme, that it is just a matter of implementing it
> on top of the SS coder/decoder? (pardon my ignorance).
>

Our wireline digital video transmission system convolved the data with a pseudo-noise data stream, converted this to analog, and put the resulting audio signal on the wire. On the receiving end, a correlation detector was used to recover the data.

In an RF system the audio would simply be shifted to the desired RF spectrum using suppressed-carrier frequency-conversion techniques. It would be interesting to consider how one might do this up-conversion (and down conversion on the receiving end) digitally. In the HF bands we're talking about signals that some DSPs could handle.

Well, more food for thought anyway &:)

Hugh

16. Re: [HFSIG:23] found CCIR 549-2

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Thu, 20 Oct 1994 09:01:39 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941020085912.29382k-100000@daffyduck>
Rendered body type: text/plain

On Wed, 19 Oct 1994, Barry Buelow - WA0RJT wrote:

>
> FREE COPIES!
> Well sort of. The first 5 people who request a copy will get one free.
> Please be able to make some use of it!
>

Dibs! And I'd be happy to reimburse you for copying and postage. Let me know how much. I'm hoping to be of assistance in designing and coding the channel simulator.

Hugh Shane
N7UAX
2915 NE 52nd St #402
Seattle, WA 98105

17. Introduction

From: wdubose@sacdm01.kelly.af.mil (WALTER (WALT) D. DUBOSE - PKT)
Date: Thu, 20 Oct 94 13:00:25 -0500
To: hfsig@tapr.org
Message-ID: <9410201800.AA16818@sacdm01.kelly.af.mil>
Rendered body type: text/plain

Greetings All,

Let me introduce myself to the "net".

I am presently employed by the USAF at Kelly AFB, San Antonio Air Logistics Center as Systems Manager for the Central Contracting Contract Data Management Host (spell that office automation). I retired from the AF Reserve in 1993 after 26 years in the reserve program. For the last 14 years in the program, I was Chief of Communications Maintenance for the 32nd Aeromedical Evacuation Group and senior communicator in the Aeromedical Evacuation System.

During Desert Shield/Storm, I was Chief of Communications for the Theater Aeromedical Evacuation System and stationed in Saudi Arabia. The Aeromedical Evacuation System used HF communications exclusively for command and control communications.

The year before Operation Just Cause, the "liberation" of Panama, the AF held an exercise which included having a NCS at Kelly AFB and an operational location in Panama. This, as we were to learn, a good practice for Operation Just Cause. During Just Cause, we learned that the 300 baud (Bell 102 Modem) communications we were using would *not* hack the communications needs for Aeromedical Evacuation (AE). We had begun using a 300 baud communications device/terminal almost 3 years before and it had connectivity problems from the first; however, it was so much better than voice (un-encrypted) communications that we put up with the downside.

Because we fell flat with AE communications during Just Cause, our Major Command arranged a test of high speed data capability using Harris Corp, HF Comm Division's MIL-STD-188-110c modems and a NSA blackbox that used the MIL-STD 39 parallel tone modem in the winter of '90 at Scott AFB. During the test we were able to pass 40K files over HF at 2400 and 4800 BPS with no trouble except for *BIG* QSBs, QRN and QRM. We would lose chunks for the file but what we received was 100% correct. No ARQ was used...just straight RATT (ASCII) with ProCom as the terminal program.

The Major Command (MAJCOM) Chief Surgeon and MAJCOM AE Commander and my unit commander directed me to include a similar demonstration/test during AE's summer 90 exercise at Ft Hunter-Liggett, CA. I arranged for Harris and SAIC/SAIT to bring equipment (the MIL-STD modems and TEMPEST Laptop computers) to demonstrate high speed HF data transmission. We were networked the Navy Hospital Ship on the west Coast, Army and Air Force Medical Units and AE units on HF and compare the capability of our old equipment (300 baud, Bell 102) with the new modems running TCP/IP.

SAIC/SAIT forgot that their software TCP/IP (probably KA9Q NOS) would be required to key the transmitter. Thus, we could only duplicate the previous demonstration but did confirm that 2400 and 4800 BPS data transmissions on HF were possible. 90 days after the exercise concluded, I was in Saudi.

While in Saudi, we had the opportunity to use Harris ALE and MIL-STD-188-110c modems. The ALE was *NOT* suitable for our network operations (that's another story) and because our AE nets were much like NTS nets, I doubt that ALE will ever work on the hambands.

HF high speed data transmissions did work and work very well. The robustness of these modems running at 2400 BPS at 100 watts provided better thruput than our 300 baud equipment running 1000 watts. Some units found that the MIL-STD modems on 100 watt transceivers worked better than 100/300 baud RATT (ASCII) at 20 KW.

As one of the evaluators of Harris' AN/URC-119 (commercially known as

the RF-350 series) HF equipment prior to production and having keep up my relationship with Harris technical personnel, I was privy to the technical information on their MIL-STD-188-110c modems. I also had the opportunity to evaluate Rockwell-Collins HF equipment that was similar to the Harris AN/URC-119 system. The MIL-STD, high speed, robust modems use external DSP technology in their modems to achieve the modem/transmission protocol.

Almost a year ago, Lou, N5SGL, discovered some of Johan's (KC7WW) work and he and I began exchanging E-Mail about robust high speed HF modems using commercial-off-the-shelf (COTS) sound boards. During my many messages with Johan, I have come to believe that hams (those of you with a much high level of technical competency than I) can reproduce the MIL-STD modem/transmission protocols using a DSP/ASP DMA soundcard.

One major question must be answered. Do we (hams) want to simply duplicate these proven MIL-STD modems (that cost a bundle) or roll our own along the same line. The MIL-STD modems BW will (just) fit into a SSB bandpass...but do we want something that wide? The will run 4800 BPS w/o the Reed-Solomon (RS) coding and 2400 with RS coding. How many BPS can we get using a 8 tone (narrow spaced tones), modem running 100 baud and can we keep the BW down to something that hams in general and the FCC will "buy"? Also, the cost of the "system" must be affordable. Should we use Golay or RS encoding for the FEC? Many questions to be answered.

I believe we can obtain 1200 BPS with FEC and ARQ/CRC and 2400 BSP w/o FEC but with ARQ/CRC in a *ROBUST* modem protocol. I truly would like to see a TCP/IP HF WAN (SMTP only) to augment and eventually replace the HF NTS nets.

73,

Walt/K5YFW

Snail-Mail Walt DuBose
 10909 Meadowhome
 San Antonio, Texas 78230

MABELL HP 210-696-3196
 BP 210-925-6081

VHF-FM* 146.46 (w/CTCSS 146.2) / 147.06 (w/CTCSS 203.5)

UHF-FM* 446.0 (w/CTCSS 146.2)

HF SSB Mobile 7.213 LSB

HF CW (40/20) U_CALL_IT

E-Mail k5yfw@sat.ampr.org (goes to k5yfw@k5yfw.ampr.org & k5yfw@sacdm10)
(ham stuff...un-official...but important)

E-Mail k5yfw@sacdm10.kelly.af.mil
(semi-official...all mailgroups including hfsig)

E-Mail wdubose@sacdm01.kelly.af.mil
(official U.S. Government stuff and emergency communications)

* Just in case you come San Antonio.

18. Re: [HFSIG:28] Re: found CCIR 549-2

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Thu, 20 Oct 1994 20:47:38 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410210147.UAA19886@allanon.ia.net>
Rendered body type: text/plain

Hey - you made the cut!

73 Barry

>
>
>
> On Wed, 19 Oct 1994, Barry Buelow - WA0RJT wrote:
>
> >

> > FREE COPIES!
> > Well sort of. The first 5 people who request a copy will get one free.
> > Please be able to make some use of it!
> >
>
> Dibs! And I'd be happy to reimburse you for copying and postage. Let me
> know how much. I'm hoping to be of assistance in designing and coding the
> channel simulator.
>
> Hugh Shane
> N7UAX
> 2915 NE 52nd St #402
> Seattle, WA 98105
>

19. Re: Z8530 and WB6YMH mod

From: Charles Brain <g4guo@dircon.co.uk>
Date: Sat, 22 Oct 1994 11:19:50 +0100
To: hfsig@tapr.org
Message-ID: <199410221019.AA01808@dircon.co.uk>
Rendered body type: text/plain

Charles Brain <g4guo@dircon.co.uk> writes:

> Hello Folks,
> I am desperately trying to obtain details os WB6YMH's 'trick' that allows
> a Z8530 SCC chip to act as a dumb serial/parallel converter. I am trying
> to take a serial synchronous bit stream from a modem and process it
> inside a P.C. !HELP!
>
> Regards Charles Brain G4GU0
>
> --
> -----
> | 73 Charles H. Brain G4GU0 |
> | E-mail g4guo@dircon.co.uk |
> | Tel Day (0245)353221 Xtn 3254 |
> | Eve (0245)469382 |
> | TCP/IP g4guo.ampr.org AX25 g4guo@gb7esx |
> | Snail 2 Daisy Court, North Springfield, Chelmsford. |
ESSEX. CM1 5QU. U.K
>

20. Re: [HFSIG:29] Introduction

From: Charles Brain <g4guo@dircon.co.uk>
Date: Sat, 22 Oct 1994 11:19:57 +0100
To: hfsig@tapr.org
Message-ID: <199410221019.AA01817@dircon.co.uk>
Rendered body type: text/plain

Hello Walter

> While in Saudi, we had the opportunity to use Harris ALE and
> MIL-STD-188-110c modems. The ALE was *NOT* suitable for our network
> operations (that's another story) and because our AE nets were much
> like NTS nets, I doubt that ALE will ever work on the hambands.
>

I would be really interested to know why you feel the ALE was unsuitable
for this type of operation. Overhere in Europe we have our doubts as well
but these are to do with the much higher levels of interference especially
during daylight hours.

Also are you talking about MIL_STD-188-141A or Harris's fast ALE system which
is being developed as part of AHFDS (Advanced HF Data System).

I believe that some form of narrower band ALE and RTCE is needed to try to spread
activity away from 20 meters and towards the WARC/higher bands.

I operate Clover from this QTH and so far have only worked stations on 20 meters
even during times when the higher bands have been open. Our H.F bands are a wonderful
resource that we are hardly even beging to use. I ask you AX25 on 20 meters!
(Oophs sorry no protocol wars!!!).

Regards Charles.

21. Re: [HFSIG:32] Re: Introduction

From: Charles Brain <g4guo@dircon.co.uk>
Date: Sat, 22 Oct 1994 12:34:29 +100 (BST)
To: hfsig@tapr.org
Message-ID: <Pine.3.89.9410221244.A5544-0100000@tdc.dircon.co.uk>

Rendered body type: text/plain

Sorry I meant night time and begining!

```
-----
| 73 Charles H. Brain G4GUO          |
| E-mail g4guo@dircon.co.uk         |
| Tel Day (0245)353221 Xtn 3254     |
| Eve (0245)469382                  |
| TCP/IP g4guo.ampr.org AX25 g4guo@gb7esx |
| Snail 2 Daisy Court, North Springfield, Chelmsford. |
| ESSEX. CM1 5QU. U.K               |
-----
```

22. Re: [HFSIG:32] Re: Introduction

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Sat, 22 Oct 94 08:42:56 CST
To: hfsig@tapr.org
Message-ID: <2111@k5yfw.ampr.org>
Rendered body type: text/plain

Greetings Charles and All,

In Charles message <199410221019.AA01817@dircon.co.uk> he writes:

> Hello Walter

>

> > While in Saudi, we had the opportunity to use Harris ALE and
> > MIL-STD-188-110c modems. The ALE was *NOT* suitable for our network
> > operations (that's another story) and because our AE nets were much
> > like NTS nets, I doubt that ALE will ever work on the hambands.

> >

>

> I would be really interested to know why you feel the ALE was unsuitable
> for this type of operation. Overhere in Europe we have our doubts as well
> but these are to do with the much higher levels of interference especially
> during daylight hours.

> Also are you talking about MIL-STD-188-141A or Harris's fast ALE system which
> is being developed as part of AHFDS (Advanced HF Data System).

> I believe that some form of narrower band ALE and RTCE is needed to try to spread
> activity away from 20 meters and towards the WARC/higher bands.

> I operate Clover from this QTH and so far have only worked stations on 20 meters
> even during times when the higher bands have been open. Our H.F bands are a wonderful
> resource that we are hardly even beging to use. I ask you AX25 on 20 meters!
> (Oophs sorry no protocol wars!!!).

>

> Regards Charles.

You have the correct MIL-STD referenced for the ALE we used.

The Areomedical Evacuation (AE) net is much like a traffic net here in the U.S. (Please note I didn't say colonies Charles as I got a personal flame from a super patriot last time I used it...Hi Hi) We had up to 30 or so stations checked into the net on voice and when they need to pass traffic, they called the net station, got permission to call another station and then passed their traffic using a Bell 103, 300 baud, terminal device. This was the normal pre-ALE method.

When we got ALEs on all our Harris HF radios, then the individual station would simply initiate an automatic connect with the station he wished to talk to and when voice contact was established, they would carry on in voice or with their terminal devices. Since there is muting on the audio in monitor mode and the ALE is scanning the 3 or 4 frequencies we were assigned, if another station needed to talk to another station, they pushed the connect button and the ALE went to the channel it had memorized as the place where the called station should be or started calling on each of the assigned/programmed channels. If on any of those channels there was a voice or data QSO going on, it would place its signaling tone on the frequency it did not recognize it and "stepped" on the QSO. If critical AE data was being passed, it had to be re-transmitted. It really had a way of ruining a QSO as the signaling took many seconds of transmission time on each frequency to establish a connection.

I am afraid that on crowded hambands, unless we have specific frequencies set aside for ALE signalling, we will incur the wrath of many hams. Also, this is compounded by the hidden transmitter effect...even if ALEs were made smart enough to listen for signals and not transmit on channels/frequencies

where a QSO existed...it might be able to hear only one side of a QSO thus stepping on the QSO.

I hope I have explained the simply and adequately enough from an operational stand point rather than any technical standpoint.

On another note, I notice that you say "Our H.F bands are a wonderful resource that we are hardly even beging to use." I did notice on several of my trips to Europe and while in Saudi that the HF bands there have *much* less activity than on this side of the pond.

Reference AX.25 on 20 meters. Other than the lack of a good modem...and Johan, KC7WW and give you information on how to build a very good Bell 102 format modem to use on AX.25. However, you must remember that you cannot realiable receive data at over 100 or so baud on HF due to the ever present mulit-path signals...this should be covered in some of HALs writings. I have summerized some of these and am willing to place them here on the SIG if there is interest.

Try going to 110 baud on AX.25 on 20 meters and see if you don't find the thruput much better. Also, run just one connected session on a frequency or limit it to *no more* than two connected sessions on a frequency. You may find the thuput comparable to 100 baud RTTY/ASCII with no errors.

73, Walt-k5yfw@home.today

PS, Charles, notice I used the "American" spelling of meters... Hi Hi. -- k5yfw

23. Re: [HFSIG:34] Re: ALE

From: Charles Brain <g4guo@dircon.co.uk>
Date: Sat, 22 Oct 1994 17:57:33 +0100
To: hfsig@tapr.org
Message-ID: <199410221657.AA04088@dircon.co.uk>
Rendered body type: text/plain
k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW) writes:
> Greetings Charles and All,

> If on any of those channels there was a
> voice or data QSO going on, it would place its signaling tone on
> the frequency it did not recognize it and "stepped" on the QSO.
> If critical AE data was being passed, it had to be
> re-transmitted. It really had a way of ruining a QSO as the
> signaling took many seconds of transmission time on each
> frequency to establish a connection.

Hi Walter,

O.K I understand about the problem, The fredricks ALE controller has a voice detect function, however as they are only using a TMS320C10 I guess it is not very effective. As you say this is a problem. I have seen an intelligent squelch that uses a DSP in conjunction with a neural net to recognise speech on a channel, it takes a few hours to train though. Possibly some form of pilot tone SSB would be more suitable. Or the ALE could send a "Is the channel free call" I guess there are a lot of diffent things that could be done.

As far as Chelmsford is concerned no I am not a member although I have been to the club a couple of times and I am friendly with one of the committe members. I will pass on your greetings.
By the way I work at Mr Guglielmo's Wireless Works in Chelmsford.

As far as packet is concerned I wonder if it would be possible to use a DSP based channel conditioner using the HDLC flags as a training sequence to reduce the multi-path problems. Also if a data frame has to be sent many times it may be possible to store multiple copies of the frame and do a majority vote on the data. It should be possible to do this without a change in the protocol. There is probably some flaw in my idea.

I do think an improved packet protocol for HF is needed, I know a lot of people are thinking about the problem. I would favour a waveform similar to CLOVER i.e using a small number of tones to reduce baud rate and a priority ack protocol.

As far as the English is concerned, my computer speaks Colonial English, anyway my father came from the Canadian Colony so I can just about understand you lot!!

Regards Charles.

24. Re: [HFSIG:35] Re: ALE

From: Jack Taylor <n7oo@huachuca-emh8.army.mil>
Date: Sat, 22 Oct 94 16:03:23 MST
To: hfsig@tapr.org
Message-ID: <m0qypYA-0000v0C@dptspd.sat.datapoint.com>
Rendered body type: text/plain

Charles Brain suggests some changes could be made to ax.25 to make it more effective for HF operations.

Be advised a revision to the ax.25 standard is "in the works" and is currently before the ARRL futures committee for review. If anyone is interested in looking over the proposed new ax.25 standard it is available via anonymous ftp from hereford.ampr.org /pub/hamradio/lapa.txt. If after review, comments wish to be made, they can be sent to one of the principle authors, bbeech@huachuca-emh8.army.mil.

73 de Jack

25. (no subject)

From: (unknown sender)
Date: (date unavailable)
Rendered body type: text/plain

Received: from Kay-Abernathy.tenet.edu by dptspd.sat.datapoint.com with smtp (Smail3.1.28.1 #3) id m0qyqVA-0001EuC; Sat, 22 Oct 94 19:09 CDT
Received: (from gjones@localhost) by Kay-Abernathy.tenet.edu (8.6.9/8.6.9) id TAA23199 for hfsig@tapr.org; Sat, 22 Oct 1994 19:09:30 -0500
From: Greg Jones <gjones@tenet.edu>
Message-Id: <199410230009.TAA23199@Kay-Abernathy.tenet.edu>
Subject: Re: [HFSIG:36] Re: ALE
To: hfsig@tapr.org
Date: Sat, 22 Oct 1994 19:09:29 -0500 (CDT)
In-Reply-To: <m0qypYA-0000v0C@dptspd.sat.datapoint.com> from "Jack Taylor" at Oct 22, 94 06:11:00 pm
X-Mailer: ELM [version 2.4 PL23]
Content-Type: text
Content-Length: 726

Also - the published doc, with the changes the committee requested should be available from the ARRL by request the first of the year.

Cheers - greg

According to Jack Taylor:

>
> Charles Brain suggests some changes could be made to ax.25 to make it more
> effective for HF operations.
>
> Be advised a revision to the ax.25 standard is "in the works" and is
> currently before the ARRL futures committee for review. If anyone is
> interested in looking over the proposed new ax.25 standard it is available
> via anonymous ftp from hereford.ampr.org /pub/hamradio/lapa.txt. If after
> review, comments wish to be made, they can be sent to one of the principle
> authors, bbeech@huachuca-emh8.army.mil.
>
> 73 de Jack
>

26. Re: [HFSIG:36] Re: ALE

From: Ron Mackay <rmackay@bud.peinet.pe.ca>
Date: Sat, 22 Oct 1994 22:35:26 -0300 (ADT)
To: hfsig@tapr.org
Message-ID: <Pine.3.89.9410222257.A12779-0100000@bud.peinet.pe.ca>
Rendered body type: text/plain

On Sat, 22 Oct 1994, Jack Taylor wrote:

> Charles Brain suggests some changes could be made to ax.25 to make it more
> effective for HF operations.
>
> Be advised a revision to the ax.25 standard is "in the works" and is
> currently before the ARRL futures committee for review. If anyone is
> interested in looking over the proposed new ax.25 standard it is available
> via anonymous ftp from hereford.ampr.org /pub/hamradio/lapa.txt. If after
> review, comments wish to be made, they can be sent to one of the principle
> authors, bbeech@huachuca-emh8.army.mil.

>
> 73 de Jack
>
Tried to get that file but "permission denied" instead for anonymous
log-in at least.

73 - Ron VE1AIC

27. ASYNC vs Coherent

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Sat, 22 Oct 1994 20:54:39 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410230154.UAA07692@allanon.ia.net>
Rendered body type: text/plain

I note that some of the comments on ALE refer to async.

This thought hasn't been a part of the discussions yet but I thought
it would be interesting before we all run off in one direction.

I'm not directly involved with the design of GPS rcvrs, but they
are getting _really_ small, low power and, most important, CHEAP.
Street prices today are still a few hundred \$. You can buy reasonable
quantities of multi-channel units for in the range of \$300 if you
know where to look.

This all leads to rather good frequency standards being possible in
the not too distant future. Now being is phase lock is a little
different issue, but real good frequency and timing are a start
in the right direction. Knowing where and when to look for a signal
is much different than just free running.

73 Barry
wa0rjt

28. Re: [HFSIG:38] Re: LAPA

From: Charles Brain <g4guo@dircon.co.uk>
Date: Sun, 23 Oct 1994 08:52:22 +0100
To: hfsig@tapr.org
Message-ID: <199410230752.AA11958@dircon.co.uk>
Rendered body type: text/plain

Ron Mackay <rmackay@bud.peinet.pe.ca> writes:
> On Sat, 22 Oct 1994, Jack Taylor wrote:
>
> Tried to get that file but "permission denied" instead for anonymous
> log-in at least.
>
> 73 - Ron VE1AIC
>
>
>

Hi Ron and all,

Yes I got the same thing!

Regards Charles

29. Re: [HFSIG:39] ASYNC vs Coherent

From: Charles Brain <g4guo@dircon.co.uk>
Date: Sun, 23 Oct 1994 08:52:17 +0100
To: hfsig@tapr.org
Message-ID: <199410230752.AA11954@dircon.co.uk>
Rendered body type: text/plain

Barry Buelow - WA0RJT <barry@ia.net> writes:
> I note that some of the comments on ALE refer to async.
>
> This thought hasn't been a part of the discussions yet but I thought
> it would be interesting before we all run off in one direction.
>

Hello Barry,

Certainly a synchronous system should reduce calling time on air. A small
on air pool of calling frequencies could be used then after contact has been
made a QSY to another frequency could be initiated to prevent blocking of the

calling channels.

Non synchronous "ALE" stations could call asynchronously.

As we are not a military system we can rely on external time sources, in the UK we have a transmission on 60 KHz called MSF this is an atomic time standard you can buy a kit of parts to build a receiver for about \$20. I guess WWV could be used in the U.S. While we wait for GPS to drop in price.

The next problem is to channelise the HAM bands to allow easier handling in the database.

Another big problem is the Radios themselves, when your rig reads say 14.070Mhz what frequency is it actually on. On SSB the dial frequency should be the frequency of the suppressed carrier. The other modes well I dont know.

Also it would be nice to disable the input filtering or have solid state switching of the filters as when my TS850 scans I hear the relays clicking (and wearing out).

Frequency stability is becoming a big issue especially on modes like CLOVER where you have to be within 10 or so Hz of the correct frequency and stay there.

It would be nice if all modern radios had an external frequency reference input.

Timeslots for sounding of stations would have to be arranged so as not to cause collisions.

The modulation scheme during calling would need to be optimised for :

- 1) Fast recognition.
- 2) Robustness
- 3) Easy to measure S/N, BER, MP etc.
- 4) Measurement of path delay (this could be used to generate simple ionograms)

A spin off of this of course would be a wonderful beacon system.

All of this is perfectly possible with modern technology but I sometimes wonder if the guys using 5 watts and a hand key don't have the right idea.

ALE systems are being developed to reduce the skills required of the operator and turn him into a simple communicator, operating is much of the fun of HAM radio.

Bring back spark!

Regards Charles

30. Official Observers

From: KY1TLuck@aol.com
Date: Sun, 23 Oct 1994 10:46:29 -0400
To: hfsig@tapr.org
Message-ID: <941023104625991923@aol.com>
Rendered body type: text/plain

I, for one, appreciate Greg's efforts at getting appropriate and detailed info regarding this particular 00/FCC matter out into the hands of those subscribed to this list. And I'm glad to see that ARRL HQ's Regulatory Information Branch continues to be so promptly responsive. This is good...

I might be able to shed some light on a couple of other aspects of this matter, since it was I who was in charge of the ARRL/FCC Amateur Auxiliary (Official Observer) program for much of my 7 years at ARRL HQ. And having been an Official Observer Coordinator in the ARRL's volunteer Field Organization in two different ARRL Sections (Virginia and Eastern Massachusetts), I have an unusual perspective from both sides of the fence.

I can understand, for example, why Fred Sober would be writing to FCC. And I can understand -- from the administrative side -- why that's a bad move on Fred's part.

For one thing, it's absolutely forbidden. I know of no less than four Official Observers and two Official Observer Coordinators who were "removed" from their positions because of doing just that. The ARRL gets real jumpy when one of their troops makes contact with FCC, and this instance we're discussing right now is absolutely a PERFECT example of why that's so. Yah, I know; we all -- most of us anyway -- tend to think that word from Johnny Johnston is somehow that of God himself, FCC-wise. But there's two things to consider carefully:

1. Johnny doesn't work for the enforcement part of FCC. In other words, while he may (and often does) express his opinion about certain Part 97 regulations, they're just that, opinions. They're not binding. Only Part 97 itself is. And let's get back to Part 97 in just a sec.
2. I've personally witnessed the results of a VERY large number of inquiries

from amateurs to FCC in the past decade. And I can assure you that, depending on who is telephoned, faxed, written to, pestered at a hamfest, or otherwise communicated with, the resulting opinions from FCC staff can -- and usually are -- remarkably different. This is clearly because of the way the question is posed, and it stands to reason that since we amateurs are typically very bad at communicating (sigh...) we're GOING to get differing opinions.

I'm getting painfully wordy, so let me sum up.

The whole issue is moot. Greg has already correctly put things in perspective; a packet bulletin is NOT, by ANY stretch of the imagination a one-way transmission. Transmissions from one TNC to another are two-way transmissions. Not one-way. Period. Thus, the supposed Notices of Apparent Liability that Fred Sober speaks of will never occur. And if 00 notices happen, that's perfectly fine since they don't have the weight of Part 97 enforcement anyway, except by way of friendly amateur-to-amateur peer pressure.

I'd hate to guess how many whining phone calls I got every day for seven years from amateurs who'd gotten "picked on" by 00 notices. My usual retort to them was "gee, have you considered throwing the post card in the trash and getting on with your amateur radio activities, OM?" "Well, (sputter, fume), but what are you gonna DO about this jerk who sent me the card, huh?" "well, I might drop him a note, congratulating him for continuing to express his opinions on what he believes would make the bands a better place to hang out". Which usually resulted in more sputtering and fuming, aimed at ME this time, which was probably better than having it aimed at the volunteer who was taking his best -- even if sometimes misguided -- shot.

And, finally, let's all try to remember that this is a non-issue. NALs won't happen, and 00 notices don't count, so let's move on.

73

Luck Hurder, KY1T ** KY1TLUCK@AOL.COM **
Supporter of AMSAT, TAPR, and NARA

31. Re: [HFSIG:40] Re: LAPA

From: Jack Taylor <n7oo@huachuca-emh8.army.mil>
Date: Sun, 23 Oct 94 8:57:40 MST
To: hfsig@tapr.org
Message-ID: <m0qz5St-0000d8C@dptspd.sat.datapoint.com>
Rendered body type: text/plain

I'd forgotten that the lapa.txt file on hereford.ampr.org was a "working copy" of the proposed revised standard for ax.25 and not available for anonymous ftp. In view of Greg's comment that the ARRL will be releasing the document (presumably for comment) in the near future, probably best to wait for that one. That way will avoid confusion as everyone will be singing "from the same songbook".
73 de Jack

32. Re: [HFSIG:42] Official Observers

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Sun, 23 Oct 1994 12:10:07 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410231710.MAA13394@allanon.ia.net>
Rendered body type: text/plain

Luck,

I appreciate your comments on the 00 and FCC. It is good to have someone with first hand experience comment.

As you may recall, I threatened to petition the FCC on exactly this issue a few months ago. At the DCC in Mpls, I attempted to defend my position that bltns are one-way and therefore restricted to info bltns.

The attitude of the sysops in the crowd was:

1. ignorance of the rules - I expected better, but wasn't totally surprised.

2. why bother - no one seems to see that there are NO values and any topic is acceptable for aa bltn.

Obviously I was disappointed at being laughed off the stage. One major technical contributor to packet said "let the channels get jammed, then there will be a reason to go faster".

The major problem I have with this entire episode is that it is one more reason that people with real brains will leave ham radio. The idiots are overrunning us and the liberals are letting them.

It is very difficult to keep from screaming "dumb shit" into the microphone.

If this is an indication of where ham radio is going, I'll go back to my model trains.

73and thanks again for valuable insights,

Barry wa0rjt

33. BBS Issue

From: Greg Jones <gjones@tenet.edu>
Date: Sun, 23 Oct 1994 15:34:47 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410232034.PAA16608@Kay-Abernathy.tenet.edu>
Rendered body type: text/plain

Hi Folks - would suggest we move the issue regarding the 00 message concerning packet BBS messages be moved over to the BBSSIG mailing list, and let the HFSIG continue messages on the current topic.(i.e. simulators and more).

By the way Johan - I think we can all agree that the current content of the HFSIG has been just great! Good Job. I really look forward to what the group will eventually produce.

Cheers - Greg

34. sorry - wrong SIG

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Sun, 23 Oct 1994 21:34:54 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410240234.VAA02872@allanon.ia.net>
Rendered body type: text/plain

Sorry group,
I must have replied to the wrong msg from Luck. I'll try to be more careful.

Barry wa0rjt

35. HFSIG on the Internet...

From: muphaus@cris.com (Marv Uphaus)
Date: Sun, 23 Oct 1994 13:13:02 -0500
To: hfsig@tapr.org
Message-ID: <kUggkCysSYs8073yn@cris.com>
Rendered body type: text/plain

Barry wrote:

>It is very difficult to keep from screaming "dumb shit" into the
>microphone.
>
>If this is an indication of where ham radio is going, I'll go
>back to my model trains.

Barry and all...! This is the reason that all this TAPR HFSIG is being done on the Internet and not on a 40 meter digital net... I have been a ham for 39 years and I have seen it getting worse each year... It's the reason that a vast majority of the intelligent hams left ham radio and went to other endeavors, the technical ones to computers... So here we are on the Internet discussing HF radio... hi hi...

Marv... K4BVG...

-- Marv Uphaus -- muphaus@cris.com -- CompuServe: 72122,1253 --
-- PGP Public Key available in plan -- finger muphaus@cris.com --
-- U.S. Mail: 4031 Airport Blvd. #49 -- Mobile, AL 36608 USA --
-- Packet Radio: K4BVG @W4IAX.#MOBAL.AL.USA.NA Ph: 205 343-9256 --

36. Re: [HFSIG:41] Re: ASYNC vs Coherent

From: wdubose@sacdm01.kelly.af.mil (WALTER (WALT) D. DUBOSE - PKT)
Date: Mon, 24 Oct 94 08:45:58 -0500
To: hfsig@tapr.org
Message-ID: <9410241346.AA28372@sacdm01.kelly.af.mil>
Rendered body type: text/plain

In Charles message of 23 Oct 1994 at 0256 CDT, he writes:
> Barry Buelow - WA0RJT <barry@ia.net> writes:
> > I note that some of the comments on ALE refer to async.
> >
> > This thought hasn't been a part of the discussions yet but I thought
> > it would be interesting before we all run off in one direction.
> >

Charles and All,

Let me address some of these comments and then I'll go back to other messages and address some others (perhaps).

I don't think the Harris, or Collins-Rockwell ALE units are concerned with being Async or Sync...perhaps I'm missing a thought or two here. The Fredericks unit is made under license from Harris thus I would assume it is basically the same unit (cheaper parts? - cheaper labor?).

>
> Hello Barry,
>
> Certainly a synchronous system should reduce calling time on air. A small
> on air pool of calling frequencies could be used then after contact has been
> made a QSY to another frequency could be initiated to prevent blocking of the
> calling channels.
> Non synchronous "ALE" stations could call asynchronously.

I believe that this would be imperative...but how many? 5, 10? How wide a channel...5, 1, 1.5, 3 KHz? Who would decide where they would be located...hams (by agreement), the FCC in the states or other regulatory agency elsewhere or perhaps the IARU? This would require an accurate receiver and ALE signaling is *not* simply a short burst.

For those not familiar with the signaling technique let me briefly describe the process in un-technical terms.

1. The signal is a low baud rate robust signal...this could be made faster.

2. Your receiver is "scanning" all the ALE channels (yes...those frequencies *would* channelize the ham bands) or only those for the mode you wished to use. You should be scanning at a specific rate and if you are scanning more than one band, you need a multi-band antenna...try this on for 40 & 20 meters.

3. The signaling transmitter must transmit a query signal on one channel for a time equal to one complete scan cycle of all the possible channels. Then it moves to the next channel and repeats the process.

Using only four frequencies (channels) in Saudi during the Desert Twins, it took more than two minutes (total cycle query time) to try and establish contact with a station if they never answered.

Can you imagine what it would be like with 10 hams on 20 meters trying to make an ALE connect on just four voice calling frequencies. *And* this does not consider what would happen if someone decided to have a QSO on the ALE channel or simply was coordinating a move to a working frequency. You *would* have to have DSP or syllabic (correct my spelling if incorrect) squelch to keep from stepping on a QSO. All of this assumes that you don't have a hidden transmitter...then the matter really gets sticky.

Calling frequencies in the U.S. have never been used as calling frequencies but rather "gathering" channels. How has this worked in other countries/parts of the world? Has it worked any better? Perhaps we in the U.S. are just not as regimented.

>
> As we are not a military system we can rely on external time sources, in the

> UK we have a transmission on 60 KHz called MSF this is an atomic time standard
> you can buy a kit of parts to build a receiver for about \$20. I guess WWV could
> be used in the U.S. While we wait for GPS to drop in price.

Why not include a separate receiver in the ALE equipment if this is needed.

>
> The next problem is to channelise the HAM bands to allow easier handling in the
> database.

Good luck on the HF bands. See above.

>
> Another big problem is the Radios themselves, when your rig reads say 14.070Mhz
> what frequency is it actually on. On SSB the dial frequency should be the
> frequency of the suppressed carrier. The other modes well I dont know.
> Also it would be nice to disable the input filtering or have solid state switching
> of the filters as when my TS850 scans I hear the relays clicking (and wearing out).

Ah yes...you *WILL* (yes I screamed) need 10 Hz resolution and accuracy and no drift and let me tell you as chief of communications maintenance for the militray using the latest state-of-the-art equipment, maintaining an accurate 10 Hz resolution drift free transceiver is *not* easy and the equipment is *expensive*. Tho, software might make this easier, you would need software that was "rig" specific and assuming all rigs could be tuned by software/hardware.

> Frequency stability is becoming a big issue especially on modes like
> CLOVER where you have to be within 10 or so Hz of the correct
> frequency and stay there.

See Above.

> It would be nice if all modern radios had an external frequency
> reference input.

This isn't really needed if the units frequency standard is accurate.

> Timeslots for sounding of stations would have to be arranged so as not to cause
> collisions.

Rgr Rgr...see above

> The modulation scheme during calling would need to be optimised for :
> 1) Fast recognition.
> 2) Robustness
> 3) Easy to measure S/N, BER, MP etc.
> 4) Measurement of path delay (this could be used to generate simple ionograms)
>

> A spin off of this of course would be a wonderful beacon system.

I think this is a better solution...a world-wide systems of beacons to automatically determine band is best with hardware and software in the users transceiver...making it passive.

All this assumes reciprocity.

> All of this is perfectly possible with modern technology but I
> sometimes wonder if the guys using 5 watts and a hand key don't have
> the right idea. ALE systems are being developed to reduce the
> skills required of the operator and turn him into a simple
> communicator, operating is much of the fun of HAM radio.

> Bring back spark!

Well, I wouldn't go that far...but basic CW is the basic digital mode.

73, Walt/K5YFW@work

37. Re: [HFSIG:35] Re: ALE

From: wdubose@sacdm01.kelly.af.mil (WALTER (WALT) D. DUBOSE - PKT)
Date: Mon, 24 Oct 94 09:16:06 -0500

To: hfsig@tapr.org
Message-ID: <9410241416.AA29710@sacdm01.kelly.af.mil>
Rendered body type: text/plain

Again, Greetings.

In Charles message of 22 Oct 1994 at 1204 CDT, he writes:

> k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW) writes:
> > Greetings Charles and All,
>
> If on any of those channels there was a
> > voice or data QSO going on, it would place its signaling tone on
> > the frequency it did not recognize it and "stepped" on the QSO.
> > If critical AE data was being passed, it had to be
> > re-transmitted. It really had a way of ruining a QSO as the
> > signaling took many seconds of transmission time on each
> > frequency to establish a connection.
>
> Hi Walter,
> O.K I understand about the problem, The fredricks ALE controller has
> a voice detect function, however as they are only using a TMS320C10
> I guess it is not very effective. As you say this is a problem. I have
> seen an intelligent squelch that uses a DSP in conjunction with a neural
> net to recognise speech on a channel, it takes a few hours to train though.
> Possibly some form of pilot tone SSB would be more suitable. Or the ALE
> could send a "Is the channel free call" I guess there are a lot of diffent
> things that could be done.

I think a sylabelic (check/correct my spelling) squelch such as
Kenwood has on one/some of their Marine HF rigs would be all that is
needed...perhaps a little more but I don't think a lot of DSP is
needed here...perhaps all the voice/data recognition could be in
one small chip.

>
> As far as Chelmsford is concerned no I am not a member although I have
> been to the club a couple of times and I am friendly with one of the
> committe members. I will pass on your greetings.
> By the way I work at Mr Guglielmo's Wireless Works in Chelmsford.

For others, I sent a message to Charles referencing my wonderful visit
to the UK several years ago and my visit with the Chelmsford ARC. If
you ever have an opportunity to visit the UK, have tea (morning,
afternoon, anytime) with the UK hams. They a most wonderful lot.
Charles works for the Macaroni Radio Works in Chelmsford...a enormous
plant with all sorts of antennae (see Charles, I can spell some things
correctly) sticking up all over the place.

> As far as packet is concerned I wonder if it would be possible to
> use a DSP based channel conditioner using the HDLC flags as a training
> sequence to reduce the multi-path problems. Also if a data frame has to be
> sent many times it may be possible to store multiple copies of the frame and
> do a majority vote on the data. It should be possible to do this without
> a change in the protocol. There is probably some flaw in my idea.

This is for programmers and technical types...please take note.

> I do think an improved packet protocol for HF is needed, I know a
> lot of people are thinking about the problem. I would favour a
> waveform similar to CLOVER i.e using a small number of tones to
> reduce baud rate and a priority ack protocol.

Phil Karn is working on a replacement to AX.25...perhaps his input is
needed here.

I don't think we want to mix a transmission protocol with a modem
protocol...I've never been comfortable with the layered structure so
I don't use it...please excuse this little quirk.

I refer to CLOVER as a modem and transmission protocol. I like to
keep them separete...this is a personal thing so just humor me.
While AX.25 may/could use improvement/change/replecement, I'd rather
focus on the modem protocol just a bit. The Bell 103 standard is a
modem protocol...CLOVER's four parallel tones is a modem protocol,
MIL-STD-188-110c (U.S. DoD) 39 parallel tone modem is a modem
protocol. Perhaps we first need to create a modem protocol/modem to
pulg onto existing transmission protocols...AMTOR, RTTY/ASCII, AX.25,
G-TOR, Pactor. Later adding the transmission protocol.

However, for the programmers, it may be easier to do both.

> As far as the English is concerned, my computer speaks Colonial
> English, anyway my father came from the Canadian Colony so I can
> just about understand you lot!!

73, Walt/K5YFW@work

38. Re: [HFSIG:39] ASYNC vs Coherent

From: "Johan Forrer"
Date: Mon, 24 Oct 1994 10:01:55 PST8PDT
To: hfsig@tapr.org
Message-ID: <5F8C9AD7E20@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Barry

You have a good point here - if we were to lay the groundwork for a new system, a synchronous approach would probably have definite advantages. (note that synchronous refers to synchronous detection methods - there also is synchronous symbol detection, that is something different).

In this regard: from what I gather, a GPS receiver would be an ideal component in a SS system. Are there any provisions for us doing direct sequence SS on HF? What would it take - STA? If we can resolve the details on resolving full monitoring capabilities of amateur radio SS on HF (which I believe is possible), this may be the answer to our problems.

Am I just day dreaming, or would it be possible to build a RF spreading/despreading subsystem, say taking a 3 Khz voice channel and utilizing a slot, say 14.000 - 14.200 Mhz?

73's Johan

39. Re: [HFSIG:49] Re: ALE

From: "Johan Forrer"
Date: Mon, 24 Oct 1994 09:36:24 PST8PDT
To: hfsig@tapr.org
Message-ID: <5F85C821A8E@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi folks,

Wow - I'm pleased to see all the activity. Good stuff!

A few points to respond to:

Sound card details:

The sound card that I'm using contains an Analog Devices ADSP-2115 DSP chip as well as the standard Windows sound chip, the AD1848. The architecture for this low cost board was developed by Analog Devices and named the "Personal Sound Architecture" (PSA). Several companies use this chip set on their sound cards, i.e., Cardinal Pro 16 (plus), Orchid Soundwave 32, Western Digital, Adaptec, Wearnes Beethoven, Echo Speech. There may be others too.

As far as compatibility is concerned, their digital interface are all the same, however, there are some differences in audio circuitry. I have some experience with the Cardinal and the Orchid Soundwave 32. They both work just fine for DSP dev. work, though are second rate sound cards if that is what you are looking for. I bought the Orchid product because it uses the latest and fastest revision of the ADSP-2115 DSP chip, i.e., 20 Mhz vs. the 16 Mhz in the others. The Cardinal Pro 16 is available as low as \$80 from PC Connection - I paid in the \$160 range for the Orchid card.

As far as DSP software is concerned - I wrote two articles (and Jon Bloom and the QEX editorial staff did a great job to make the materials look very professional) about this card: The August 1994 issue deals with the hardware registers and how to program it using a free software toolkit - the second one will appear in the upcoming (November 1994) QEX. It shows the implementation of an adaptive 100/200 baud DSP HF modem. I think Marv will enjoy that one as it will answer most of his questions.

The PSA architecture was designed with the upcoming Windows 95 (Chicago) DSP API in mind - In the mean time, Analog Devices are giving away free SDK for using this card doing DSP development work under Windows. I have

written a few programs using this platform - however, it requires yet another layer of programming and further knowledge of yet another programming philosophy to get even something simple working. It does work reasonably well. I do believe, however, that for our purposes, we would need to stick to writing in assembly and DOS (for a while at least).

Other software that I have available for these PSA cards:

- * PSATOR - AMTOR for the PSA cards (RTTY/ASCII will be added in future)
- * PSA-Pactor - Pactor for PSA cards. This is a full rx/tx implementation using the adaptive modems mentioned above. It also implements full 16-bit memory ARQ in conjunction with brute force search algorithms. A 386/25 or better computer is required.
- * LMS noise reduction - A port of the W9GR code.

These are shareware and I have attempted to keep the UCSD ftp site up to date. However, I will provide copies if you could be so kind to send me a formatted disk and SASE/mailler.

I hope this brief reply answers the questions on hardware / software for the sound card. There are further software tools available, but I don't want to waste further bandwidth on that for the moment.

HF Channel simulator

Lets wait till we have had a chance to study the materials. I think Jon and Hugh both indicated interest in implementation details, so I am sure that we can start exchanging ideas about that shortly.

ALE performance

I think I understand that the present implementation of ALE has both a routing function as well as a messaging structure. What interests me about this protocol, is its modulation scheme - it uses 8-ary FSK and Golay (24,12) coding. It would be relatively easy to implement for amateur radio application - just a thought: perhaps replacing 300 baud packet?

The future of AX.25

Although it may be a while until we progress to the data link layer, I do think it would be wise if we pay some attention to these efforts. Please keep us posted on what is happening - thanks.

73's

Johan

40. Re: TENTATIVE PROJECT OUTLINE

From: Charles Brain <g4guo@dircon.co.uk>
Date: Mon, 24 Oct 1994 18:54:42 GMT
To: hfsig@tapr.org
Message-ID: <199410241854.AA06286@dircon.co.uk>
Rendered body type: text/plain

"Johan Forrer
<FORRERJ@frl.orst.edu> writes:

> Hi All,
>
>
> I trust that you have had an opportunity to look at the modulation
> schemes in Table 1. of my last posting. I thought it would be of
> interest to provide further outlines of those ideas and get some
> discussion and interaction started. Please be so kind and take a
> few moments to study the summary given below.
>
> Remember, the HFSIG can handle multiple active threads, however,
> please be sure to use an appropriate "subject" when you post your
> replies. This way we will know what your message is about.
>
Hello Johan et al,

FL"

I joined the net after you posted this could you email it to me or re-post it please.

I was thinking if we use these modems in the rtty sections of the band we have a shroud idea of the QRM we are going to see.

- 1) 170Hz FSK
- 2) 200Hz FSK
- 3) CW
- 4) 4 tone ALE
- 5) Ourselves!

Regards Charles.

41. Re: [HFSIG:48] Re: ASYNC vs Coherent

From: Charles Brain <g4guo@dircon.co.uk>
Date: Mon, 24 Oct 1994 18:54:37 GMT
To: hfsig@tapr.org
Message-ID: <199410241854.AA06273@dircon.co.uk>
Rendered body type: text/plain

wdubose@sacdm01.kelly.af.mil (WALTER (WALT) D. DUBOSE - PKT) writes:
>
>
> In Charles message of 23 Oct 1994 at 0256 CDT, he writes:
> > Barry Buelow - WA0RJT <barry@ia.net> writes:
> > > I note that some of the comments on ALE refer to async.
> > >
> > > This thought hasn't been a part of the discussions yet but I thought
> > > it would be interesting before we all run off in one direction.
> > >
>
> Charles and All,
>
> Let me address some of these comments and then I'll go back to other
> messages and address some others (perhaps).
>
> I don't think the Harris, or Collins-Rockwell ALE units are concerned
> with being Async or Sync...perhaps I'm missing a thought or two
> here. The Fredericks unit is made under license from Harris thus I
> would assume it is basically the same unit (cheaper parts? - cheaper
> labor?).
>

Hi Walt,

The sync part I was referring to was at call setup. All stations in the net that are able to scan are listening on the same channel at any time. This means there is no scanning section to the call. An originating station can estimate the best channel tune and wait, at the right moment when everyone is listening it makes the call. This also means that everyone is listening at the most likely time to hear someone else.

Net synchronisation is vital, in an amateur environment an external sync source can be used i.e GPS. In a military environment the network has to self sync. Normally this consists of starting in async operation, when one hears a more senior node in the net that has sync you sync to him etc.

However when the net starts to get busy performance drops off dramatically. Sync systems normally have about 100 channels in the scan group, also every few minutes a new set of 100 channels are allocated. I have a paper written by a Prof Carlson in Sweden on this topic.

Regards Charles.

42. Re: [HFSIG:51] Re: ALE

From: Charles Brain <g4guo@dircon.co.uk>
Date: Mon, 24 Oct 1994 19:04:23 GMT
To: hfsig@tapr.org
Message-ID: <199410241904.AA08195@dircon.co.uk>
Rendered body type: text/plain

"Johan Forrer

FL"

<FORRERJ@frl.orst.edu> writes:

> ALE performance

> -----

>

> I think I understand that the present implementation of ALE has both a
> routing function as well as a messaging structure. What interest me about
> this protocol, is its modulation scheme - it uses 8-ary FSK and Golay
> (24,12) coding. It would be relatively easy to implement for amateur
> radio application - just a thought: perhaps replacing 300 baud

> packet?

Hello again,

It needs tidying up but I have written some C code that does GOLAY (24,12) as per the MIL STD encoding and decoding. It uses a look-up table to do the encoding and also another look up that takes the error syndrome and finds the error vector which you XOR with the original data, it also tells you how many errors it has detected so you can either correct or detect. It seems to work i.e it corrects errors O.K. I used the Parity matrix in the standard.

Regards Charles

43. Re: [HFSIG:52] Re: TENTATIVE PROJECT OUTLINE

From: Greg Jones <gjones@tenet.edu>
Date: Mon, 24 Oct 1994 14:33:19 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410241933.0AA22723@Kay-Abernathy.tenet.edu>
Rendered body type: text/plain

Just as a reminder, all messages from any of the TAPR SIGs are available by month from the listserv.

Simply send a message to listserv@tapr.org and in the message body state 'index -all' or in this case 'index hfsig'.

This will then get you a listing of the files that can be requested.

The command for that is:

get area filename

Example:
get tapr taprinfo.txt

This would get you the file taprinfo.txt in the area tapr. When you do the index, you will see that the files you want are in an area, so when you do the get, just fillin the area and filename.

This way, anyone can review the entire SIG from the start.

Cheers - Greg

President -- Tucson Amateur Packet Radio Corp

TAPR Office (817) 383-0000 | Internet: gjones@tenet.edu

According to Charles Brain:

>
> "Johan Forrer
<FORRERJ@frl.orst.edu> writes:
> > Hi All,
> >
> >
> > I trust that you have had an opportunity to look at the modulation
> > schemes in Table 1. of my last posting. I thought it would be of
> > interest to provide further outlines of those ideas and get some
> > discussion and interaction started. Please be so kind and take a
> > few moments to study the summary given below.
> >
> > Remember, the HFSIG can handle multiple active threads, however,
> > please be sure to use an appropriate "subject" when you post your
> > replies. This way we will know what your message is about.
> >
> Hello Johan et al,
>
> I joined the net after you posted this could you email it to me or
> re-post it please.
>
> I was thinking if we use these modems in the rtty sections of the band
> we have a shroud idea of the QRM we are going to see.
>
> 1) 170Hz FSK
> 2) 200Hz FSK
> 3) CW

FL"

> 4) 4 tone ALE
> 5) Ourselves!
>
> Regards Charles.
>

44. Re: [HFSIG:53] Re: ASYNC vs Coherent

From: wdubose@sacdm01.kelly.af.mil (WALTER (WALT) D. DUBOSE - PKT)
Date: Mon, 24 Oct 94 14:37:11 -0500
To: hfsig@tapr.org
Message-ID: <9410241937.AA07884@sacdm01.kelly.af.mil>
Rendered body type: text/plain

Charles,

In your message of 24 Oct 1994 at 1404 CDT, you write:

> wdubose@sacdm01.kelly.af.mil (WALTER (WALT) D. DUBOSE - PKT) writes:
> > Charles and All,
> >
> > Let me address some of these comments and then I'll go back to other
> > messages and address some others (perhaps).
> >
> > I don't think the Harris, or Collins-Rockwell ALE units are concerned
> > with being Async or Sync...perhaps I'm missing a thought or two
> > here. The Fredericks unit is made under license from Harris thus I
> > would assume it is basically the same unit (cheaper parts? - cheaper
> > labor?).
> >
> >
> > Hi Walt,
> > The sync part I was refering to was at call setup. All stations in
> > the net that are able to scan are listening on the same channel at any
> > time. This means there is no scanning section to the call. An originating
> > station can estimate the best channel tune and wait, at the right moment
> > when everyone is listening it makes the call. This also means that everyone
> > is listening at the most likely time to hear someone else.
> > Net synchronisation is vital, in an amateur environment an external sync
> > source can be used i.e GPS. In a military environment the network has to
> > self sync. Normally this consists of starting in async operation, when one
> > hears a more senior node in the net that has sync you sync to him etc.
> > However when the net starts to get busy performance drops off dramatically.
> > Sync systems normally have about 100 channels in the scan group, also every few
> > minutes a new set of 100 channels are allocated. I have a paper written by
> > a Prof Carlson in Sweden on this topic.

I'd really hate to tie my HF operation to the presence of a satellite...
If I'm going to do that, I might as well get a nitch on a geosync
bird and send 19.2 data and to heck with HF. Besides will I now have
to have two receivers, coax and antennas just to work HF. And too,
my truck is already filled with radio equipment and antennas..I will
now need another "rig" and antenna on/in the truck and will have to
park in the hot sun to work HF...not under a cool shade tree cause I
gotta receive the satellite.

The reason many U.S. military units use HF is they don't want to
depend on a satellite...their *too easy* to knock out.

Realizing that GPS boxes are and will come down in price, and I'll
need another box for the ALE, I'll soon have a bunch of money in
equipment.

If we don't watch out, an HF rig will cost 2000 PS.

I might add...where's the KISS?

Walt/KSYFW

45. Re: [HFSIG:50] Re: ASYNC vs Coherent

From: Barry Buelow - WA0RJT <barry@ia.net>
Date: Mon, 24 Oct 1994 22:53:36 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410250353.WAA12461@allanon.ia.net>
Rendered body type: text/plain

>
> Hi Barry
>
> You have a good point here - if we were to lay the groundwork for a new

> system, a synchronous approach would probably have definite
> advantages. (note that synchronous refers to synchronous detection methods -
> there also is synchronous symbol detection, that is something different).

exactly what I was thinking. tx and rx in at least freq sync and possibly
in phase sync.

>
> In this regard: from what I gather, a GPS receiver would be an ideal
> component in a SS system. Are there any provisions for us doing direct
> sequence SS on HF? What would it take - STA? If we can resolve the details
> on resolving full monitoring capabilities of amateur radio SS on HF (which
> I believe is possible), this may be the answer to our problems.
>

It seems to me that any serious improvement in hf throughput is not
going to come from an inexpensive chrome box that plugs in to the mic
and speaker jacks.

There is a cascading effect. Good throughput requires good freq/time
stability. 99.99% of off the shelf rigs do NOT support external freq
references.

Without addressing SS for now, this leads to a total system design:
timebase, tx, rx, modem. Consider that a good system out to be
something in the 50 to 100W range, and doesn't need all the chrome
and knobs of typical modern radios, single band or a couple of bands,
what else?

This is a rather substantial design task, but it brings all the important
technical issues into play. Major flexibility of operation issues are
not a design requirement.

If this system cost \$1000, it would be a huge bargain.

> Am I just day dreaming, or would it be possible to build a
> RF spreading/despeading subsystem, say taking a 3 KHz voice channel and
> utilizing a slot, say 14.000 - 14.200 Mhz?
>
> 73's Johan
>
Just dreaming along...

73 Barry

46. Re: [HFSIG:51] Re: PSA

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Tue, 25 Oct 1994 10:03:26 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941025091408.15133A-100000@daffyduck>
Rendered body type: text/plain

On Mon, 24 Oct 1994, Johan Forrer FL wrote:

>
> As far as compatibility is concerned, their digital interface are all the
> same, however, there are some differences in audio circuitry. I have some
> experience with the Cardinal and the Orchid Soundwave 32. They both work
> just fine for DSP dev. work, though are second rate sound cards if that is
> what you are looking for. I bought the Orchid product because it uses the

Is the audio interface on these cards sufficient for digital communication?

>
> The PSA architecture was designed with the upcoming Windows 95 (Chicago)
> DSP API in mind - In the mean time, Analog Devices are giving away free SDK
> for using this card doing DSP development work under Windows. I have
> written a few programs using this platform - however, it requires yet
> another layer of programming and further knowledge of yet another
> programming philosophy to get even something simple working. It
> does work reasonably well. I do believe, however, that for our purposes, we
> would need to stick to writing in assembly and DOS (for a while at least).
>

How about Linux? With it's multitasking capabilities, Linux is an excellent
platform for packet. And, of course, for software development you can't
beat Unix. All we need are Linux driver for the Personal Sound
Architecture and Unix-hosted development tools.

> Other software that I have available for these PSA cards:

>
> * PSATOR - AMTOR for the PSA cards (RTTY/ASCII will be added in future)
> * PSA-Pactor - Pactor for PSA cards. This is a full rx/tx implementation
> using the adaptive modems mentioned above. It also implements
> full 16-bit memory ARQ in conjunction with brute force search
> algorithms. A 386/25 or better computer is required.
> * LMS noise reduction - A port of the W9GR code.

How about K9NG/G3RUH modulation? (I'm also putting together a node for our local VHF TCP/IP network.)

73
Hugh

47. Re: [HFSIG:58] Re: PSA

From: "Johan Forrer"
Date: Wed, 26 Oct 1994 08:37:40 PST8PDT
To: hfsig@tapr.org
Message-ID: <62763272701@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hugh,

My experiences with the audio of the sound card tells me that its more than adequate for our usage. The modems that I have played with only runs at a minimal sampling rate of 5.5125 kHz and I have found the anti-aliasing/recon filters very effective to be able to run at that low sample rate. The LMS noise reducer runs at about 19 kHz and the audio quality is quite good, if not better than the commercial boxes doing the same thing. The only thing that the card does not have is digital I/O. However, someone with a bit of ingenuity can fabricate a "port". I use a serial port on the host for that purpose in the interim.

Tom, HB9JNX has written 1200 and 9600 baud packet modems for the card - I have no further details except knowing of its exsistance. I do know, however, that he told me that he built a direct interface to the DSP's "SPORT" for data I/O. So - it can be done. Tom has been very quiet lately, so I dont know the status of that project of his. It would be real nice if we can make a 9600/G3RUH modem available for the card. Perhaps you should be the one do it?

I hope to post a summary on materials that I have been working on regarding modulation schemes and possible ways of debugging/evaluating the system, by monday.

Please give your thoughts on your DSP platform high priority - the learning curve is real steep and we will be starting to do some real stuff in the very near future. Let me assure you - there is a lot of fun ahead!

Hope I addressed all the questions.

73's

Johan

48. Re: [HFSIG:58] Re: PSA

From: "Johan Forrer"
Date: Wed, 26 Oct 1994 08:52:11 PST8PDT
To: hfsig@tapr.org
Message-ID: <627A1243BB7@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hugh,

About Linux - I have a system that is used for experimentation. And I do think that it has tremendous capabilities, unfortunately, however, it is a very fast moving target and I have found myself maintaining the OS more than doing any exiting DSP development work. A DSP platform for Linux will probably require a kernel hack similar to the AX.25 project for Linux - You really have to know what you are doing. I do believe that one could probably put something really nice together.

On the other hand - if you are interested in wider audience, Linux is somewhat out on a limb. There really is only one fellow doing the sound drivers (and he has quit doing that), and from what I gathered, it is quite generic - a far cry from trying to actually run your own DSP code on your sound card. On the other side of the coin, with the battle going on between the Microsoft and OS/2 worlds, something useful may emerge. In following the

DSP API developments for these new OS's it appears there are some good things in stall - of course not for ham radio, but for multimedia running on DSP's. Fortunately we will be able to fit right into the picture with our work. So have good faith.

Sorry for the distraction - this has little to do with HF digital but thought I would give my two cents worth.

--Johan

49. Re: [HFSIG:59] Re: PSA

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Wed, 26 Oct 1994 10:39:00 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941026101729.15133C-100000@daffyduck>
Rendered body type: text/plain

> lately, so I dont know the status of that project of his. It would be real
> nice if we can make a 9600/G3RUH modem available for the card. Perhaps you
> should be the one do it?

This would be a tasty project indeed! Is this the appropriate place to begin a discussion on this topic, or is there another SIG that would be better.

>
> Please give your thoughts on your DSP platform high priority - the learning
> curve is real steep and we will be starting to do some real stuff in the
> very near future. Let me assure you - there is a lot of fun ahead!
>

In comparing DSP platforms I consider performance, features, compatibility, price, and availability. IMHO, for our purposes, there are only two contenders for a DSP platform at this time: the PSA-class boards and the TAPR DSP-93. The PSA solution has adequate performance, is reasonably priced and available now. However, it only works with ISA/EISA machines and is not code compatible with the TAPR product, which has a large code base that is certain to grow in the future. On the other hand, the TAPR board is twice as expensive as the PSA boards and has a lead time of half a year.

If I didn't care about what the rest of the ham radio world was doing I'd go with the PSA solution. But I *do* care! Does it make sense to not take advantage of a potentially huge pool of code? Or maybe we should just start our own faction which will have its own devotees and its own code base. Maybe in time this will become the winning solution due to the low price and general availability of the hardware.

I think I'll continue to sit on the fence a little while longer. Can someone push me off?

Hugh

50. Re: [HFSIG:61] Re: PSA

From: "Johan Forrer
Date: Wed, 26 Oct 1994 11:40:01 PST8PDT
To: hfsig@tapr.org
Message-ID: <62A6D8F00FF@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hugh,

I see no problem discussing the 9600 baud modem stuff here, though think that if it concerns implementation details, the DSP development SIG would be the right place and may appeal to a wider audience. I think you will get a lot more support there too.

In a future posting (hopefully by monday), I hope to include some important aspects that differentiates the VHF style DSP modems from the intended robust HF DSP modems. They have lots of similarities. i.e. phase and symbol synchronous, carrier & data extractors from suppressed-carrier modulation systems, however, require careful design of signal constellation, and particulaly, robust phase locking - these are much more important than speed. But more on this later.

Johan

51. Re: [HFSIG:61] Re: PSA

From: Greg Jones <gjones@tenet.edu>
Date: Wed, 26 Oct 1994 14:44:21 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410261944.OAA09751@Kay-Abernathy.tenet.edu>
Rendered body type: text/plain

According to Hugh Shane:

> > lately, so I dont know the status of that project of his. It would be real
> > nice if we can make a 9600/G3RUH modem available for the card. Perhaps you
> > should be the one do it?
>
> This would be a tasty project indeed! Is this the appropriate place to
> begin a discussion on this topic, or is there another SIG that would be
> better.

I think you will find that the cards do not have any enough power to handle the 9600 baud full-duplex transmission. From what I have been told it is doubtfull that half-duplex can be done..maybe somebody can reply to this.

This does not stop doing a lower baud with more bps, but current standards favor FSK, for a lot of reasons. I would point you at some of the last TAPR prcoeedings and ARRL DCC articles on 2 and 4 FSK and PSK and the like performance possibilities.

> > Please give your thoughts on your DSP platform high priority - the learning
> > curve is real steep and we will be starting to do some real stuff in the
> > very near future. Let me assure you - there is a lot of fun ahead!
>
> In comparing DSP platforms I consider performance, features,
> compatibility, price, and availability. IMHO, for our purposes, there are
> only two contenders for a DSP platform at this time: the PSA-class boards
> and the TAPR DSP-93. The PSA solution has adequate performance, is
> reasonably priced and available now. However, it only works with ISA/EISA
> machines and is not code compatible with the TAPR product, which has a
> large code base that is certain to grow in the future. On the other hand,
> the TAPR board is twice as expensive as the PSA boards and has a lead time
> of half a year.

With hope, a mfg will pick up the DSP-93 design and mfg it for much less than what we are doing it for as kits. The reason for doing the DSP-93 is not to make money at selling kits, but to open up the technology (which has been the goal since 1988). With the DSP-12 from LL Grace leaving the market that now leaves the AEA units as the only commerically available unit (from amateur suppliers).

I don't know if you can remember back to the TNC-2 introduction, but those kits were sold for \$250+ and within two years were available for \$150. We hope that the DSP-93 technology can happen in the same way. If the unit is done as all SMA technology and as one board and is done outside the US, I figure that cost would a lot less than what we can do it in small qtys and as kits.

So - if you like the design, and dont want to either 1) wait for a kit and 2) spend \$430 (which is a bargain for what it does) then you should be calling up your favored mfg and asking him when they are going to talk to tapr about OEMing the technology :-)

> If I didn't care about what the rest of the ham radio world was doing I'd
> go with the PSA solution. But I *do* care! Does it make sense to not take
> advantage of a potentially huge pool of code? Or maybe we should just
> start our own faction which will have its own devotees and its own code
> base. Maybe in time this will become the winning solution due to the low
> price and general availability of the hardware.

I think you will see that code is developed for all platforms and a lot of it will be ported between systems. The reason for going with one solution or another is what you want to do. If you need two radio ports, the interface to the radios, and processor and power to do 9600 baud full-duplex or the like then the DSP-93 is a solution.

If you want to do modulations that require the A/D power but not all the processor power than a sound card is a good solution. Just don't forget that with the sound card you still need all the radio interface depending on what you are wanting to do...that is one reason the DSP-93 is more expensive to start with - it has all the interface built in.

> I think I'll continue to sit on the fence a little while longer. Can someone
> push me off?

Probably not a bad decision. If you have a modem you are happy with now, then you should stick with it. You will want to purchase a DSP unit if 1) you want to experiment and 2) there is a new mode that requires it. No need to purchase something that will sit around.

Cheers - Greg

52. Re: [HFSIG:63] Re: PSA

From: "Johan Forrer"
Date: Wed, 26 Oct 1994 13:01:31 PST8PDT
To: hfsig@tapr.org
Message-ID: <62BC9234604@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Greg,

Could you please post the actual reference to DCC and TAPR docs that have the FSK/PSK performance specs for everyone's benefit - would be great - thanks.

-Johan

53. DEMODULATOR TESTING ETC.

From: "Johan Forrer"
Date: Thu, 27 Oct 1994 12:53:49 PST8PDT
To: HFSIG@tapr.org
Message-ID: <643A91E04AC@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

Thanks for the words of encouragement - the HFSIG is only what we make it to be and it is heartwarming to see that there still is amateur spirit to push the envelope.

Re: the DSP platform: I hope we don't waste too much time on a silly issue of who has more marbles. The DSP soundcard will do 9600 baud - I don't know who told Greg otherwise, just to put the record straight - look at the Scout 14.4K modems, they use the same family DSP chip. Trying to compare these platforms are like comparing a Caddy to a Jeep - both will do 90 on the freeway, but they are ultimately for different purposes. The DSP-93 has excellent capabilities - especially a capable strong support group - that is vital if you are just starting out - lots of "free" software - and it could be used as a free-standing DSP box. I hope they get all the support to make their efforts worthwhile - that box must succeed.

However, for those that are going to become deeply involved in the development and testing aspects, the DSP development platform requirements are going to become much more complex - for example, the real-time channel simulator probably will be running on its own DSP processor with input - output channels to take the signal from the modulator, add the simulated effects, and play it out in real time on the output channel to the demodulator under test. This probably means a DSP platform for the modulator and another for the demodulator. Add to that DSP diagnostic tools, i.e. to display parts of the signal or constellations, logging of results etc.... That adds up to two PC platforms and three DSP development systems. Perhaps you will now understand why I am using \$80 sound DSP sound cards. If anyone can suggest another approach, I'll be really interested.

However, if you are only planning to participate in actual over the air testing, a modest PC, i.e. 486/33 and either a DSP-93 or DSP sound card will do just fine. At this level, I am confident that code for either approach will be made available for you to participate in these tests.

By Monday I'll put forward a rough proposal on something that we can start working on - unfortunately we do not have anything yet on the simulator, but I have good faith that we can put something very rudimentary together that will do for starters. If we keep things generic, and write algorithms in pseudo-code, it really won't matter what platform you are going to use at this early stage. Actually, if we follow this approach, you can even write the test code in "C" and run it on your PC - it will be a lot slower and you won't get the same statistical sampling, but all we need to establish initially are the working tolerances for some of the major components. Hopefully things will be a lot clearer after seeing the plan.

I also encourage folks to discuss other topics besides the digital issues - please feel free to do so. Also if you have other ideas on how to go about this journey - let us have your opinions/suggestions.

73's

Johan

54. Re: [HFSIG:65] DEMODULATOR TESTING ETC.

From: muphaus@cris.com (Marv Uphaus)
Date: Thu, 27 Oct 1994 20:38:29 -0500
To: hfsig@tapr.org
Message-ID: <L05ikCysS-LT073yn@cris.com>
Rendered body type: text/plain

>However, if you are only planning to participate in actual over the air
>testing, a modest PC, i.e. 486/33 and either a DSP-93 or DSP sound card
>will do just fine.

Now, Johan...

Only a few months ago an 8088 would have been a "modest PC"...
hahaha... My, how our perspective changes...!!!

>I also encourage folks to discuss other topics besides the digital issues -

See above... hahaha...

Marv...

-- Marv Uphaus -- muphaus@cris.com -- Ph: 205-343-9256 --
-- PGP Public Key available -- "finger muphaus@cris.com" --
-- U.S.Mail: 4031 Airport Blvd. #49 -- Mobile, AL 36608 --
-- Packet Radio Address -- K4BVG @W4IAX.#MOBAL.AL.USA.NA --

55. Re: [HFSIG:65] DEMODULATOR TESTING ETC.

From: Greg Jones <gjones@tenet.edu>
Date: Fri, 28 Oct 1994 12:55:59 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199410281756.MAA14326@Kay-Abernathy.tenet.edu>
Rendered body type: text/plain

According to Johan Forrer

FL:

> Re: the DSP platform: I hope we dont waste too much time on a silly issue
> of who has more marbles. The DSP soundcard will do 9600 baud - I dont know
> who told Greg otherwise, just to put the record straight - look at the
> Scout 14.4K modems, they use the same family DSP chip. Trying to compare
> these platforms are like comparing a Caddy to a Jeep - both will do 90 on
> the freeway, but they are ultimately for different purposes. The DSP-93 has
> excellent capabilities - especially a capable strong support group - that
> is vital if you are just starting out - lots of "free" software - and
> it could be used as a free-standing DSP box. I hope they get all the support
> to make their efforts worthwhile - that box must succeed.

I would be real interested in knowing if anyone has 9600 baud FSK
implemented on a card device. From my conversations with crystal
semiconductor, makers of several of the devices for the sound card industry,
they felt that 9600 baud was outside the processing power of the many of the
current low-cost devices.

Please keep in mind that 14.4Kbps is 1200 baud with 12 bits of encoding.
The problem we have in amateur radio applications with expanding the numbers
of bits per baud is the medium we are using - radio. Radio and telephonic
operations are considerably different and the S/N must be very good in
order to be able to handle the higher order encoding methods.

One thing we should keep in mind is the work Phil Karn is doing on using FEC
and symbol rates in such a way to provide some rather robust links.

Both Phil's and Tom's papers on these topics were published in
the TAPR 1994 Proceedings, which I think sells for \$7. I am always
forgetting what stuff goes for.

I think we agree that none of us should be getting into pissing contests on
performance issues. The idea of any discussion, I would hope within TAPR,
would be to do the research, make the informationl available, and then try
to implement it in something that provides the widest possible access to
amateurs in the community.

The other main reason for not really worrying about performance issues is
that whatever we have now - will be 2+ folds more powerful in 5 years.

Cheers - Greg

56. RE: [HFSIG:65] DEMODULATOR TESTING ETC.

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Fri, 28 Oct 94 09:42:00 EDT
To: hfsig <hfsig@tapr.org>
Message-ID: <2EB1002A@arrl.org>
Rendered body type: text/plain

Johan says:

> If we keep things generic, and write algorithms in
> pseudo-code, it really won't matter what platform you are going to use at
> this early stage. Actually, if we follow this approach, you can even write
> the test code in "C" and run it on your PC - it will be a lot slower and
> you won't get the same statistical sampling, but all we need to establish
> initially are the working tolerances for some of the major components.

This is what I'm most interested in: developing and documenting the algorithms needed to implement all the neat things people are dreaming up, including the HF channel simulator and faster HF modems, but not limited to those applications. The algorithms can be documented in pseudo-code, flow charts, SDL or whatever seems appropriate. Then the algorithm can be implemented on multiple platforms. My feeling is that we should strive, where possible, to do work that is platform independent, then transfer that work to an implementation on a specific platform. If we do a good algorithm design up front, it will shortly appear as an implementation on several platforms; there are enough programmers to make that happen.

And, of course, I'd like to see the algorithms documented in QEX! :-)

Regarding the HF channel simulator... its main use (for us) is to test other DSP modems, so clearly it has to be running on a separate platform. The TI DSK strikes me as a good, cheap stand-alone platform. But, as I say, if we get the algorithms documented, the choice of platform is easier.

-- Jon, KE3Z

57. Re: [HFSIG:67] Re: DEMODULATOR TESTING ETC.

From: "Johan Forrer
Date: Fri, 28 Oct 1994 16:03:07 PST8PDT
To: hfsig@tapr.org
Message-ID: <65ED1C42400@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Greg,

You quite correctly pointed out the challenge of 9600 baud FSK - I do agree that what we presently have, would not have enough ticks to do that - thanks for clearing that up.

I would like you to consider some ideas on how we can achieve reasonable bit rates based on low baud rates and multiple phase / multiple amplitude modulation, i.e. complex modulation. This would be within the capabilities of what we have right now. We need however, address the variabilities of the HF channel and its affects on such a scheme. I'll put out some ideas on Monday on the possible use of a "pilot" channel that goes out in parallel with the data channel to aid in phase/amplitude equalization for such complex modulated schemes. This is not a unique idea either.

Oh yes, quite true - coding theory will come in a bit later in the game, but should not be a bandaid for a poor modulation, it should buy us that few extra bits "gain" on top of what we have achieved with a good modulation scheme.

73's

Johan

58. HF Modulation

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Fri, 28 Oct 94 20:19:23 CST
To: hfsig@tapr.org
Message-ID: <2123@k5yfw.ampr.org>
Rendered body type: text/plain

Greetings HFers,

I thought I'd try and re-name this thread from DEMODULATION to Modulation. Realizing however, that both are needed and in complex

modems and both creating the modulator and detector on a sound card is going to be a challenge.

For those who have gotten in "late", Johan and I have been discussing robust, high speed modems for almost a year and Greg and I have been carrying on a dialog for several months now.

I have used commercial/military equipment on HF at 2400 and 4800 BPS that were *very* robust modems. We even ran on-the-air tests against SITOR (the commercial version of AMTOR) over the same channel. The high speed modems beat SITOR in transferring data at a factor greater than the difference in bit rate. That is to say that at 2400 BPS for the robust HF modem we had 100% thruput at 2400 BPS. It was capable of 2400 BPS data transfer rate and it transferred 2400 bps. On SITOR, we had less than 100% of the possible thruput rate. In Jan of 1990, the modem cost \$15,000. Today you can get it for \$3,200.

I have seen the block diagram and chips it has and it "looks" like a sound card and has the same type DSP & ASP chips as soundcards. While probably would not want to just emulate that modem, we should improve on the design, we should make it a goal as we know it is achievable.

Many of my friends ask me why I hate CLOVER. I tell them that I don't but feel for that amount of money, hams should get more BPS for their buck. Hear's the reason. CLOVER uses 4 parallel tones to get 800-900 BPS. The commercial/military modems use 39 tones to get 2400 BPS with the 40th tone as an unmodulated/standard modulated pilot/doppler tone. I believe that some where between 4 and 40 tones hams (those of you who are much smarter than I) can come up with a modem for HF that will be very robust and produce 1200-2400 (and maybe more) BPS in a BW that is less than the commercial/military modem.

If the guys who will do the actual programming "max-out" the soundcard and produce a robust, 2400 BPS (100 baud or less) HF modem and can apply the the same principle to a V/UHF modem and get 9600/19.2K BPS, ham radio will have greatly benefited. If they can produce only 1200 BPS on HF and 4800 or 9600 on V/UHF, thats Ok too...just max-out the sound cards capability. By then, perhaps better sound cards and/or other devices will be on the market that will let us go higher.

I like the picture of the frog choking the water fowl while the frog's head in in the bird's mouth. The caption is never give up. Don't give up, try hard, keep it low cost and KISS.

73 & have a nice weekend ya'll.

Walt@home (dba k5yfw)

59. Modulation

From: Johan Forrer <70730.3472@compuserve.com>
Date: 31 Oct 94 00:33:51 EST
To: HFSIG <HFSIG@tapr.org>
Message-ID: <941031053350_70730.3472_CHK45-1@CompuServe.COM>
Rendered body type: text/plain

Hi all,

It is not quite so obvious which "magic" technology we should pursue that will give maximum gains. Well, perhaps there is no magic answer, or at least; those that know better won't say either or won't be bothered. Lets face it, most everything that we are about to embark on, have been done before in one form or another commercially - though I am not aware of anything affordable for amateurs to experiment with (yes, Clover could have been the answer, except that it was developed as a "closed" architecture).

How one chooses one modulation/demodulation scheme over another, is an extensive and interesting topic by itself. Without going into much details, let me take an easy way out and go by what others have achieved, that is, m-FSK to approx 300 bps and (nxm)-PSK for higher rates. These have been found to be quite usable on HF.

The m-FSK option, i.e. a system similar in modulation and coding than what is presently implemented for ALE, would be quite do-able on our present DSP platforms. Anyone interested in that?

Nxm-PSK is a multitone approach. In this scheme, there are several (n- of them) independent m-PSK carriers. As far as

development is concerned, one can concentrate your development efforts on getting the most out of one carrier, then add additional carriers as far as your DSP horsepower will take you, or perhaps until you run out of bandwidth (suspect the DSP is the limiting factor). The parallel deal is to be used to get maximal bit rate as conditions permit, or serve for additional redundancy when conditions are not so good. This basically is the premise used for the MIL-STD-188 parallel modems. I would (rather bravely) like to suggest that this alternative be pursued - unless of course you could suggest a better alternative. Please let me know your opinion on this one.

Assuming the nxm-PSK approach in its simplest form, how does it sound if we develop a single m-PSK channel and see how far that takes us. ?

I would suggest as background, obtaining the following three papers (these are heavy duty, but quite good):

1) Foschini, G.J., R.D. Gitlin, and S.B. Weinstein. "On the Selection of a Two-Dimensional Signal Constellation in the Presence of Phase Jitter and Gaussian noise." Bell System Tech. Jour. Vol. 52(6) July-August 1973 pp:927-965.

2) Simon, M.K., and W.C. Lindsey. "Optimum Performance of Suppressed Carrier Receivers with Costas Loop Tracking". IEEE Trans. on Communications Vol. COM-25(2) February 1977 pp:215-227.

A more complete and realistic treatment of the performance of different modulation schemes and of phase lock loops in the presence of noise and phase jitter are also given in the above authors' book: "Telecommunications Systems Engineering." Dover pubs. ISBN 0-486-66838-X.

3) Simon, M.K. and J.G. Smith. "Carrier Synchronization and Detection of QASK Signal Sets." IEEE Trans. on Communications Vol. COM-22(2) February 1974.

It is evident, at least to me, that carrier phase synchronization and amplitude equalization becomes quite important when dealing with these complex (meaning quadrature) waveforms. The idea of using a separate "pilot" channel that carries such phasing and clock data, seems rather attractive. Not only will it simplify implementation details, but probably allow us to be able to track multipath effects. The additional robustness in system performance that it would add, would offset its small amount of additional bandwidth. Let me have your views on this one too. Also think a bit about what information need to be encoded in the "pilot" carrier.

73's

Johan

PS: Thanks to Walt for forwarding a wealth of details over an extended period on the MIL-STD-188 modems. We certainly have beaten its possibilities to death. The use of the pilot tone, or doppler channel as they call it, is used in both the 39-tone as well as the 16-tone modems.

I must also thank Bill Coleman, AA4LR, for discussions on TELEBIT's PEP, and the use of the pilot tone for dealing with multipath effects.

November 1994

Original source file: hfsig.9411

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1994-11/TAPR-HF_1994-11.txt

1. Re: [HFSIG:70] HF Modulation

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Mon, 31 Oct 94 13:15:46 CST
To: hfsig@tapr.org
Message-ID: <9410311915.AA00711@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Many of my friends ask me why I hate CLOVER. I tell them that I don't
> but feel for that amount of money, hams should get more BPS for their
> buck. Hear's the reason. CLOVER uses 4 parallel tones to get 800-900
> BPS. The commercial/military modems use 39 tones to get 2400 BPS with
> the 40th tone as an unmodulated/standard modulated pilot/doppler tone.
> I believe that some where between 4 and 40 tones hams (those of you who
> are much smarter than I) can come up with a modem for HF that will be
> very robust and produce 1200-2400 (and maybe more) BPS in a BW that is
> less than the commercial/military modem.
>
> If the guys who will do the actual programming "max-out" the soundcard
> and produce a robust, 2400 BPS (100 baud or less) HF modem and can apply
> the the same principle to a V/UHF modem and get 9600/19.2K BPS, ham
> radio will have greatly benefited. If they can produce only 1200 BPS on
> HF and 4800 or 9600 on V/UHF, thats Ok too...just max-out the sound
> cards capability. By then, perhaps better sound cards and/or other
> devices will be on the market that will let us go higher.
...
> Walt@home (dba k5yfw)
>

Perhaps I'm missing a point here, or someone can correct me if I'm wrong ... I thought that there was a 500 hz bandwidth limit within the CW/RTTY segments on HF - and that one of the reasons that CLOVER is only 4 tones (carriers) is so that it would fit within that BW limit.

- Tom, N5EG

2. Re: [HFSIG:72] Re: HF Modulation

From: "Johan Forrer"
Date: Tue, 1 Nov 1994 09:56:58 PST8PDT
To: hfsig@tapr.org
Message-ID: <6B8BA2E5D76@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Tom,

Re: 500 Hz BW limitation - our digital activies on HF are regulated by Part 97 of the FCC regulations: it says that we can use up to 300 baud or 1000 Hz shift.

The 500 Hz BW issue has been brought up recently by parts of the digital community that has actually gone so far as to file petition requesting such BW for for semi-automatic operations (mainly APlink usage). Except for the Clover modulation scheme, none of the *presently available* equipment is capable of staying within 500 Hz. This is due to the "fuzzy" definition of how folks measure emitted BW. If you do like the Clover folks and specify BW at say 50 dB down from the main lobe, then just about everyone else uses in excess of 800 Hz. So, asking for 500 Hz BW right now with what is on the market right now, Clover is the only one that will comply for APlink usage. Does folks realize what this petition means?

Thanks for bringing up this point - hopefully we will show that with the right kind of technology, our present BW allocation can go a long way towards high speed HF digital - there is no need for additional regulations.

73's

Johan

3. RE: [HFSIG:73] Re: HF Modulation

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
Date: Tue, 01 Nov 94 13:15:00 EST
To: hfsig <hfsig@tapr.org>
Message-ID: <2EB68625@arrl.org>
Rendered body type: text/plain

>The 500 Hz BW issue has been brought up recently by parts of the digital
>community that has actually gone so far as to file petition requesting
>such BW for for semi-automatic operations (mainly APlink usage). Except for

>the Clover modulation scheme, none of the *presently available* equipment
>is capable of staying within 500 Hz. This is due to the "fuzzy" definition
>of how folks measure emitted BW. If you do like the Clover folks and
>specify BW at say 50 dB down from the main lobe, then just about everyone
>else uses in excess of 800 Hz. So, asking for 500 Hz BW right now with what

>is on the market right now, Clover is the only one that will comply for
>APlink usage. Does folks realize what this petition means?

The specification of occupied bandwidth is in Sec 2.202 of FCC Rules and Regulations (Appendix 10 in ARRL's FCC Rule Book). It is set by the points in the spectrum where the sum of the power at frequencies above and below those points is 0.5% of the total power. (Interestingly, there is a qualification of this rule for multi-channel frequency division systems, which may have difficulty meeting such a definition.) In any case, 170-Hz shift, 100-baud transmissions (AMTOR or RTTY) with proper key shaping fall within a 500-Hz bandwidth by the above definition. The 500-Hz bandwidth limitation does *not* rule out the use of modes other than Clover!

That said, the FCC's definition of bandwidth is probably not the one we want to key on. The reason Clover strives to maintain almost all of its energy within a 500-Hz bandwidth is not for legal requirements, rather it's because doing so means you can actually use 500-Hz channelization. Try running a mediocre AMTOR link when there's another AMTOR station 500-Hz away that is 40 dB above the station you're trying to copy. That adjacent-channel station may be "occupying" a 500-Hz bandwidth, but you'll still find your copy corrupted by his "insignificant" sidebands only 30 dB or so down from his carrier.

-- Jon, KE3Z

4. HF Channel simulator

From: "Johan Forrer
Date: Tue, 1 Nov 1994 11:36:56 PST8PDT
To: HFSIG@tapr.org
Message-ID: <6BB64D6637A@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Greetings,

For those interested in the channel simulator, do yourself a favour and also obtain the Watterson et al. paper:

Clack C. Watterson, John R. Jurosheck, and William D. Bensema:
"Experimental Confirmation of an HF Channel Model". IEEE Trans. on Comm.
Tech. Vol COM-18(6) December 1970.

It appears quite doable - perhaps we should form a working group of those that wish to actively participate in coding this model? Please e-mail me directly. Does this sound reasonable? We will post our progress (if any).

73's

Johan

5. HF Modulation

From: "Johan Forrer
Date: Tue, 8 Nov 1994 09:47:30 PST8PDT
To: HFSIG@tapr.org
Message-ID: <761961E4752@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Folks,

Quiet on the homefront?

A few general comments: I have suggested some reading materials -
If you have do not have access to these, perhaps I can help make

photocopies - please e-mail me directly. These papers are quite old, though very appropriate. Unfortunately they also are quite mathematically inclined - however, don't be intimidated by a bit of algebra. If you are not interested in all the gory details, just make sure you understand the premise and note the conclusions. Some things take a while to sink in and as one develops your grasp of the subject, it becomes clearer later.

The insight that these papers attempt to provide is: given some model of the "channel", how does different modulation schemes stack up? Typically various forms of OOK (on-off keying), FSK, and PSK, are considered against white (additive Gaussian white) noise - where it is shown that: for the same amount of emitted energy, phase modulation schemes have slightly lower error rates. Now keep in mind that some rather important assumptions are made, i.e. that we have the RX and TX in perfect sync as far as carrier phase and symbol sync is concerned. Well, what happens if that is not the case and what if the channel model includes phase jitter? We see that performance drops rather dramatically. This is where the issue of robustness comes in and where we see that noncoherent FSK actually does very well - it requires little or no assumptions! However, just keep those theoretical rankings in mind as the limits of performance under ideal conditions.

Well, then why don't we just go ahead and use noncoherent FSK? Lets say we are aiming for 1200 bps (uncoded). Theoretically, this would require a minimum of 1200 Hz shift, and need at least an additional 1200 Hz to accommodate the modulation envelope - total about 2400 Hz. Two major problems: that's a lot of bandwidth for HF, second, such short bit times would be a disaster on HF with multipath and the other kinds of channel anomalies. OK - so what else can we do? We have to make a tradeoff: lower robustness for more bandwidth efficiency. That is why some of those papers looked at those complex modulated schemes, i.e. QAM etc.. and their performance under non-ideal conditions. The Foschini paper surprisingly shows that QAM does better than expected - which is great news.

The other papers is about carrier and data recovery using phase lock loops - more about this later.

One last note: the 2-3 dB loss in performance associated with a bandwidth-efficient modulation scheme, may be offset with coding gain achieved by coding theory, i.e. trellis modulation/viterbi demodulation etc. This sophistication, unfortunately, adds greater complexity to the implementation. The DSP/host algorithms are necessarily going to be closely coupled and this alone hints that an involved system is in your future.

A bit of progress with the PSK modem - the modulator is working nicely and soon I'll be testing with two sound cards back-to-back. This setup, however, is very experimental and is, for me at least, a way of learning how all these intricate algorithms work together.

I have collected and studied some further materials on the channel simulator, enough to see that it is quite doable. More about that later.

I would like to hear any further thoughts or suggestions from your side. Please let us hear from you.

73's

Johan

6. SOUNDWAVE TOOLS

From: "Johan Forrer
Date: Tue, 8 Nov 1994 10:16:12 PST8PDT
To: HFSIG@tapr.org
Message-ID: <76210DD4A6A@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

>Hi Johan,
>

>I just received my Soundwave card yesterday. It sure is fun having a new
>toy! You once mentioned (I think) that there is a freeware assembler for
>the DSP floating around out there. Can you tell me where? Also, is there
>any board-level technical documentation (i.e. block diagram, memory map,
>etc.) available to developers? I can get data books on the individual

>chips from their respective vendors, but it would be nice to know how they
>go together!
>
>Maybe you should post your reply to the net.
>
> 73s
>
> Hugh

Hi Hugh,

OK - now the fun really starts!

First get hold of the August 1994 QEX. I have described the chip set, memory layout, and mayor PC-DSP programming issues in some detail in that article.

You also *must* get the AD1848 data sheet for sure, and the 21xx user's guide for sure. That will help you with the 2115's architecture and programmer's model.

The most recent PD toolkit for the sound card is available from ftp.ucsd.edu in /hamradio/packet/tcpip/incoming/psatool1.zip, also at ftp.cs.buffalo. You will also find some further example applications code for the QEX article(s) there too. Also check out my port of the W9GR DSP for the sound card - I recently uploaded that.

I wonder if you have obtained HB9JNX's 9600 baud G3RUH modem for the sound card? If so, perhaps you could make that available too? That may be and interesting application to explore.

I would also suggest browsing Analog Devices ftp site: ftp.analog.com (137.71.23.11) for all sorts of useful bits of hardware/software information and code examples.

Keep in mind that there are *real* ADI software tools available - I recently got their assembler/linker/simulator and C-compiler at very reasonable cost during a "special" offer deal.

This will keep you busy for a while, but let me know if you need further assistance.

73's

Johan

7. Re: [HFSIG:76] HF Modulation

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Tue, 8 Nov 94 15:05:56 CST
To: hfsig@tapr.org
Message-ID: <9411082105.AA06303@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Hi Folks,
>
> Quiet on the homefront?
>
> A few general comments: I have suggested some reading materials -
> If you have do not have access to these, perhaps I can help make
> photocopies - please e-mail me directly. These papers are quite
> old, though very appropriate. Unfortunately they also are quite
> mathematically inclined - however, don't be intimidated by a bit
> of algebra. If you are not interested in all the gory details,
> just make sure you understand the premise and note the
> conclusions. Some things take a while to sink in and as one
> develop your grasp of the subject, it becomes clearer later.

....

> I would like to hear any further thoughts or suggestions from your side.
> Please let us hear from you.
>
> 73's
>
> Johan
>

Johan, I acquired a copy of "Experimental Confirmation of an HF Channel Model" IEEE ComTech, COM-18 No. 6. The channel model proposed seems

straight-forward, however the derivation of the baseband channel modulation functions, $G_i(t)$ is anything but straight-forward, and certainly a little bit beyond algebra.

Mapping of the parameters from tables I and II into deterministic formulas has so far eluded me. Have you looked into this? My guess would be to choose random processes with gaussian distributions to fit the observed data. However, I see the problem as one of mapping random PHASE of $G(t)$ into a gaussian-shaped FREQUENCY shift of the signal, a bit beyond my mathematical background at present. Do you know of a simple algorithm that generates gaussian-distributed random numbers?

Additionally, the author states that the fading of each individual propagation component is represented by a Rayleigh distribution. I think the formula for that is fairly tractable. I would assume that a random-number generator that creates a Rayleigh distribution applied to the magnitude of each $G_i(t)$ would be used here. I found a simple formula from some microwave radio stuff done here about 10 years ago (where a Rayleigh path is the standard assumption) but it is not clear what the coefficients should be after looking at Tables I and II.

The two data points were taken at 9.26 Mhz, and 5.86 Mhz. Would the data points taken from a higher frequency (14, 21, 28 Mhz) be significantly different (with a potentially longer path, one might think so).

The article is definitely interesting, but after two complete read throughs, it will still require significant study to grasp.

- Tom, N5EG

8. Re: [HFSIG:78] Re: HF Modulation

From: "Glen Worstell" <Glen_Worstell@notes.seagate.com>
Date: Tue Nov 08 16:19:03 1994
To: hfsig@tapr.org
Message-ID: <2ebfeb48.seagate@notes.seagate.com>
Rendered body type: text/plain

> Do you know of a simple algorithm that generates gaussian-distributed random numbers?

Two answers:

- 1) use the "randn" function in Matlab.
- 2) From "Communication Formulas and Algorithms", C. Rorabaugh:

- a) Generate a pseudorandom floating point number $U1$ greater than 0 and ≤ 1.0
- b) Generate another one, $U2$.
- c) $G1 = \cos(U2) * \sqrt{-2 * \ln(U1) * \sigma * \sigma}$
- d) $G2 = \sin(U2) * \sqrt{-2 * \ln(U1) * \sigma * \sigma}$

$G1$ and $G2$ will be independent zero-mean gaussian "random numbers" with standard deviation of σ .

The book gives another method which is a bit faster as it does not use trig functions.

Important: you must have a good random number generator routine to start with. some unix systems have 16 bit generators which are not very good. Also, for this sort of work the gaussian generator must have a good "crest factor". That is, the percent of numbers exceeding, say, 6 sigma must be in the ballpark of the predicted amount. This will not happen if the quality of the random number generator is not good (for example, not enuf bits).

I used this method in a disk channel simulator, with good results. Disk drives are a lot like communication channels!

Cheers,
Glen Worstell, KG0T
--
Glen Worstell -- Glen_Worstell@notes.seagate.com

Seagate Technology - 920 Disc Drive - Scotts Valley, CA 95066 USA
Main Phone 408-438-6550 - Email Problems postmaster@notes.seagate.com
Technical Support: BBS 408-438-8771 Fax 408-438-8137 Voice 408-438-8222

OGATE Version 8 message trace and attachment information:

MsgFileName: m:\mgate\outbound\516.MSG
Org Date: 11-08-94 03:20:47 PM
From: Glen Worstell@SEAGATE
To: hfsig @ tapr.org @ INTERNET
Subject: Re: [HFSIG:78] Re: HF Modulation
Attachments: none

9. Re: [HFSIG:78] Re: HF Modulation

From: "Johan Forrer"
Date: Tue, 8 Nov 1994 16:07:13 PST8PDT
To: hfsig@tapr.org
Message-ID: <767EB0E2676@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

>> Hi Folks,
>>
>> Quiet on the homefront?
>>
>> A few general comments: I have suggested some reading materials -
>> If you have do not have access to these, perhaps I can help make
>> photocopies - please e-mail me directly. These papers are quite
>> old, though very appropriate. Unfortunately they also are quite
>> mathematically inclined - however, don't be intimidated by a bit
>> of algebra. If you are not interested in all the gory details,
>> just make sure you understand the premise and note the
>> conclusions. Some things take a while to sink in and as one
>> develop your grasp of the subject, it becomes clearer later.
>
>
>
>> I would like to hear any further thoughts or suggestions from your side.
>> Please let us hear from you.
>>
>> 73's
>>
>>Johan
>>

> Johan, I acquired a copy of "Experimental Confirmation of an HF
>Channel Model" IEEE ComTech, COM-18 No. 6. The channel model proposed
>seems straight-forward, however the derivation of the baseband channel
>modulation functions, $G_i(t)$ is anything but straight-forward, and certainly
>a little bit beyond algebra.

Outstanding Tom! agree that nothing about this is that simple - but
folks should also not feel intimidated by scholarly papers. I'm a
mediocre mathematician at best with self-taught interests in DSP and COMM
theory - programming is what I like best.

Lets explore your questions and perhaps we can unravel this thing
together

>>Mapping of the parameters from tables I and II into deterministic
>formulas has so far eluded me. Have you looked into this? My guess would
>be to choose random processes with gaussian distributions to fit the
>observed data. However, I see the problem as one of mapping random PHASE
>of $G(t)$ into a gaussian-shaped FREQUENCY shift of the signal, a bit beyond
> my mathematical background at present. Do you know of a simple algorithm
>that generates gaussian-distributed random numbers?

I think you have a good grasp of what is required. What might help
further now is to get CCIR Report 549-2 - look at p.60-62 for a fuller
description of the Gaussian-scatter model and formulae. Tables I and II of
the Watterson paper is very complete and you probably don't want to try and
emulate that in its entirety. If you further obtain CCIR 520- 1, they
list all the parameters of interest for "Good", "Moderate", "Poor", and
"Flutter fading" that may be put to good use in the model.

About the random Gaussian number generator, lets keep in
mind that the baseband signal is complex, thus you need to generate a
complex random number too, i.e.:

```
WGN_real = std * sqrt(-2*ln(ran1)*cos(2*pi*ran2))  
WGN_imag = .....sin.....
```

where: std is the std. dev. of GN,

ran1, ran2 is uniformly distributed between 0,1

Then you have to think a little about how to balance the summation node, i.e. the amount of power that each tap is going to contribute.

Also think a bit about what you are going to do about the different sample rates involved at each tap, i.e. they will be somewhat different thus you would have to include interpolating filters before mixing in the doppler signal. OK?

>
> Additionally, the author states that the fading of each individual
>propagation component is represented by a Rayleigh distribution. I think
>the formula for that is fairly tractable. I would assume that a random-

Yes, that is described in the references of CCIR 549-2 and also in some detail on p.62 of 549-2.

.....< >

> The two data points were taken at 9.26 Mhz, and 5.86 Mhz. Would
>the data points taken from a higher frequency (14, 21, 28 Mhz) be
>significantly different (with a potentially longer path, one might think
>so).
>

Certainly - different frequencies will behave differently - from what I have read on this subject, multipath, is strongly dependant on the MUF.

> The article is definitely interesting, but after two complete read
>throughs, it will still require significant study to grasp.
>
> - Tom, N5EG

Appreciate your feedback. I would not have known anything on this topic were it not for generous help from others (I still am struggling with this too).

Keep us posted on your progress.

Johan, KC7WW

10. RE: HF Channel Simulator

From: Eric E Silbaugh <esilbaug@afit.af.mil>
Date: Tue, 8 Nov 1994 21:06:15 -0500
To: hfsig@tapr.org
Cc: esilbaug@afit.af.mil
Message-ID: <199411090206.AA11090@stealth.afit.af.mil>
Rendered body type: text/plain

Those of you who have been discussing HF channel simulation may be interested in the following paper. I ran across it while doing research on an unrelated topic.

L. Ehrman, L.B. Bates, J.F. Eschile, J.M. Kates, "Realtime Software Simulation of the HF Radio Channel" IEEE Trans. on Communications, August 1982, p.1809

It contains an overview of the theoretical HF channel model and details on how it was implemented on a signal processor. They used special purpose hardware but today's DSP chips should be up to the task.

For those of you interested in how various modulation methods hold up under interference, check out this paper.

O.A.H. Shabsigh, "On the Effects of CW Interference on MSK Signal Reception" IEEE Trans. on Communications, August 1982, p.1925

Wouldn't want our fancy new demodulator to crumble when some lid starts doing 5 WPM on top of our signal would we.

I have enjoyed lurking on this SIG for a few weeks. Lots of

Eric

11. Message not deliverable

About the random Gaussian number generator, lets keep in

mind that the baseband signal is complex, thus you need to generate a complex random number too, i.e.:

```
WGN_real = std * sqrt (-2*ln(ran1)*cos(2*pi*ran2))
WGN_imag = .....sin.....
```

where: std is the std. dev. of GN,
ran1, ran2 is uniformly distributed between 0,1

Then you have to think a little about how to balance the summation node, i.e. the amount of power that each tap is going to contribute.

Also think a bit about what you are going to do about the different sample rates involved at each tap, i.e. they will be somewhat different thus you would have to include interpolating filters before mixing in the doppler signal. OK?

>
> Additionally, the author states that the fading of each individual
> propagation component is represented by a Rayleigh distribution. I think
> the formula for that is fairly tractable. I would assume that a random-

Yes, that is described in the references of CCIR 549-2 and also in some detail on p.62 of 549-2.

.....< >

> The two data points were taken at 9.26 Mhz, and 5.86 Mhz. Would
> the data points taken from a higher frequency (14, 21, 28 Mhz) be
> significantly different (with a potentially longer path, one might think
> so).
>

Certainly - different frequencies will behave differently - from what I have read on this subject, multipath, is strongly dependant on the MUF.

> The article is definitely interesting, but after two complete read
> throughs, it will still require significant study to grasp.
>
> - Tom, N5EG

Appreciate your feedback. I would not have known anything on this topic were it not for generous help from others (I still am struggling with this too).

Keep us posted on your progress.

Johan, KC7WW

12. Message not deliverable

From: Administrator_at_SPAWAR-MC4@smtp-gw.spawar.navy.mil
Date: Wed, 09 Nov 94 01:00:13 EST
To: hfsig@tapr.org
Message-ID: <9410097843.AA784371667@smtp-gw.spawar.navy.mil>
Rendered body type: text/plain

>> Hi Folks,
>>
>> Quiet on the homefront?
>>
>> A few general comments: I have suggested some reading materials -
>> If you have do not have access to these, perhaps I can help make
>> photocopies - please e-mail me directly. These papers are quite
>> old, though very appropriate. Unfortunately they also are quite
>> mathematically inclined - however, don't be intimidated by a bit
>> of algebra. If you are not interested in all the gory details,
>> just make sure you understand the premise and note the
>> conclusions. Some things take a while to sink in and as one
>> develop your grasp of the subject, it becomes clearer later.
>
>
>
>> I would like to hear any further thoughts or suggestions from your side.
>> Please let us hear from you.

>>
>> 73's
>>
>>Johan
>>

> Johan, I acquired a copy of "Experimental Confirmation of an HF
>Channel Model" IEEE ComTech, COM-18 No. 6. The channel model proposed
>seems straight-forward, however the derivation of the baseband channel
>modulation functions, $G(t)$ is anything but straight-forward, and certainly
>a little bit beyond algebra.

Outstanding Tom! agree that nothing about this is that simple - but
folks should also not feel intimidated by scholarly papers. I'm a
mediocre mathematician at best with self-taught interests in DSP and COMM
theory - programming is what I like best.

Lets explore your questions and perhaps we can unravel this thing
together

>>Mapping of the parameters from tables I and II into deterministic
>formulas has so far eluded me. Have you looked into this? My guess would
>be to choose random processes with gaussian distributions to fit the
>observed data. However, I see the problem as one of mapping random PHASE
>of $G(t)$ into a gaussian-shaped FREQUENCY shift of the signal, a bit beyond
> my mathematical background at present. Do you know of a simple algorithm
>that generates gaussian-distributed random numbers?

I think you have a good grasp of what is required. What might help
further now is to get CCIR Report 549-2 - look at p.60-62 for a fuller
description of the Gaussian-scatter model and formulae. Tables I and II of
the Watterson paper is very complete and you probably don't want to try and
emulate that in its entirety. If you further obtain CCIR 520- 1, they
list all the parameters of interest for "Good", "Moderate", "Poor", and
"Flutter fading" that may be put to good use in the model.

About the random Gaussian number generator, lets keep in
mind that the baseband signal is complex, thus you need to generate a
complex random number too, i.e.:

```
WGN_real = std * sqrt (-2*ln(ran1)*cos(2*pi*ran2))  
WGN_imag = .....sin.....
```

where: std is the std. dev. of GN,
 ran1, ran2 is uniformly distributed between 0,1

Then you have to think a little about how to balance the summation node,
i.e. the amount of power that each tap is going to contribute.

Also think a bit about what you are going to do about the different
sample rates involved at each tap, i.e. they will be somewhat different
thus you would have to include interpolating filters before mixing in the
doppler signal. OK?

>
> Additionally, the author states that the fading of each individual
>propagation component is represented by a Rayleigh distribution. I think
>the formula for that is fairly tractable. I would assume that a random-

Yes, that is described in the references of CCIR 549-2 and also in some
detail on p.62 of 549-2.

.....< >

> The two data points were taken at 9.26 Mhz, and 5.86 Mhz. Would
>the data points taken from a higher frequency (14, 21, 28 Mhz) be
>significantly different (with a potentially longer path, one might think
>so).
>

Certainly - different frequencies will behave differently - from
what I have read on this subject, multipath, is strongly dependant
on the MUF.

> The article is definitely interesting, but after two complete read
>throughs, it will still require significant study to grasp.
>
> - Tom, N5EG

Appreciate your feedback. I would not have known anything on this topic
were it not for generous help from others (I still am struggling with this
too).

Keep us posted on your progress.

Johan, KC7WW

13. Message not deliverable

From: Administrator_at_SPANAR-MC4@smtp-gw.spawar.navy.mil
Date: Wed, 09 Nov 94 06:10:30 EST
To: hfsig@tapr.org
Message-ID: <9410097843.AA784390258@smtp-gw.spawar.navy.mil>
Rendered body type: text/plain

```
>> Hi Folks,  
>>  
>> Quiet on the homefront?  
>>  
>> A few general comments: I have suggested some reading materials -  
>> If you have do not have access to these, perhaps I can help make  
>> photocopies - please e-mail me directly. These papers are quite  
>> old, though very appropriate. Unfortunately they also are quite  
>> mathematically inclined - however, don't be intimidated by a bit  
>> of algebra. If you are not interested in all the gory details,  
>> just make sure you understand the premise and note the  
>> conclusions. Some things take a while to sink in and as one  
>> develop your grasp of the subject, it becomes clearer later.  
>  
> ....  
>  
>> I would like to hear any further thoughts or suggestions from your side.  
>> Please let us hear from you.  
>>  
>> 73's  
>>  
>>Johan  
>>  
  
> Johan, I acquired a copy of "Experimental Confirmation of an HF  
>Channel Model" IEEE ComTech, COM-18 No. 6. The channel model proposed  
>seems straight-forward, however the derivation of the baseband channel  
>modulation functions,  $G_i(t)$  is anything but straight-forward, and certainly  
>a little bit beyond algebra.
```

Outstanding Tom! agree that nothing about this is that simple - but
folks should also not feel intimidated by scholarly papers. I'm a
mediocre mathematician at best with self-taught interests in DSP and COMM
theory - programming is what I like best.

Lets explore your questions and perhaps we can unravel this thing
together

```
>>Mapping of the parameters from tables I and II into deterministic  
>formulas has so far eluded me. Have you looked into this? My guess would  
>be to choose random processes with gaussian distributions to fit the  
>observed data. However, I see the problem as one of mapping random PHASE  
>of  $G(t)$  into a gaussian-shaped FREQUENCY shift of the signal, a bit beyond  
> my mathematical background at present. Do you know of a simple algorithm  
>that generates gaussian-distributed random numbers?
```

I think you have a good grasp of what is required. What might help
further now is to get CCIR Report 549-2 - look at p.60-62 for a fuller
description of the Gaussian-scatter model and formulae. Tables I and II of
the Watterson paper is very complete and you probably don't want to try and
emulate that in its entirety. If you further obtain CCIR 520- 1, they
list all the parameters of interest for "Good", "Moderate", "Poor", and
"Flutter fading" that may be put to good use in the model.

About the random Gaussian number generator, lets keep in mind that the baseband signal is complex, thus you need to generate a complex random number too, i.e.:

```
WGN_real = std * sqrt (-2*ln(ran1)*cos(2*pi*ran2))
WGN_imag = .....sin.....
```

where: std is the std. dev. of GN,
ran1, ran2 is uniformly distributed between 0,1

Then you have to think a little about how to balance the summation node, i.e. the amount of power that each tap is going to contribute.

Also think a bit about what you are going to do about the different sample rates involved at each tap, i.e. they will be somewhat different thus you would have to include interpolating filters before mixing in the doppler signal. OK?

>
> Additionally, the author states that the fading of each individual
>propagation component is represented by a Rayleigh distribution. I think
>the formula for that is fairly tractable. I would assume that a random-

Yes, that is described in the references of CCIR 549-2 and also in some detail on p.62 of 549-2.

.....< >

> The two data points were taken at 9.26 Mhz, and 5.86 Mhz. Would
>the data points taken from a higher frequency (14, 21, 28 Mhz) be
>significantly different (with a potentially longer path, one might think
>so).
>

Certainly - different frequencies will behave differently - from what I have read on this subject, multipath, is strongly dependant on the MUF.

> The article is definitely interesting, but after two complete read
>throughs, it will still require significant study to grasp.
>
> - Tom, N5EG

Appreciate your feedback. I would not have known anything on this topic were it not for generous help from others (I still am struggling with this too).

Keep us posted on your progress.

Johan, KC7WW

14. Re: [HFSIG:80] Re: HF Modulation

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Wed, 9 Nov 94 07:53:43 CST
To: hfsig@tapr.org
Message-ID: <9411091353.AA06918@eagle.aud.alcatel.com>
Rendered body type: text/plain

> About the random Gaussian number generator, lets keep in
> mind that the baseband signal is complex, thus you need to generate a
> complex random number too, i.e.:
>
> WGN_real = std * sqrt (-2*ln(ran1)*cos(2*pi*ran2))
> WGN_imag = ...sin.....
>
> where: std is the std. dev. of GN,
> ran1, ran2 is uniformly distributed between 0,1
>
> Johan, KC7WW
>

Johan: the author states that the phase is uniformly distributed.
>From this one would guess that the uniform phase results in gaussian frequency, but I have a hard time believing this, actually (doesn't seem

intuitive). Given the Rayleigh amplitude distribution, I was assuming that the two components of $G_i(t)$ would be magnitude and phase, with the above two (non-complex) random distributions. Then of course we would do polar-to-rectangular conversion in order to multiply $G_i(t)$ by the delayed signal. My guess is that the tapped delay line only contains the real part of the input signal, but after multiplication by $G_i(t)$, of course it is then complex.

> Those of you who have been discussing HF channel simulation may
 > be interested in the following paper. I ran across it while doing
 > research on an unrelated topic.
 >
 > L. Ehrman, L.B. Bates, J.F. Eschile, J.M. Kates, "Realtime
 > Software Simulation of the HF Radio Channel" IEEE Trans. on
 > Communications, August 1982, p.1809
 >
 > It contains an overview of the theoretical HF channel model and
 > details on how it was implemented on a signal processor. They
 > used special purpose hardware but today's DSP chips should be up
 > to the task.
 >
 > Eric

Thanks for the reference, Eric! Nothing like seeing a real implementation.

- Tom, N5EG

15. RE: [HFSIG:85] Re: HF Modulation

From: "Bloom, Jon, KE3Z" <jbloom@arrl.org>
 Date: Wed, 09 Nov 94 10:30:00 EST
 To: hfsig <hfsig@tapr.org>
 Message-ID: <2EC0EAF@arrl.org>
 Rendered body type: text/plain

I just acquired a copy of "Simulation of Communication Systems," by Michel C. Jeruchim, Philip Balaban and K. Sam Shanmugan. (1992, Plenum Press, New York. ISBN 0-306-43989-1.) If you can find a copy of this (I bought mine, to the tune of \$110), you'll find more than you ever wanted to know about this subject! I've yet to find time to delve into it deeply, but it looks like it covers the subject in detail, including providing the mathematical background.

```
>> WGN_real = std * sqrt (-2*ln(ran1)*cos(2*pi*ran2))
>> WGN_imag = .....Sin.....
>>
>> where: std is the std. dev. of GN,
>>       ran1, ran2 is uniformly distributed between 0,1
```

Should the trig functions be inside the square root as shown? In the book referenced above, (in section 3.11, "Computer Generation of Random Numbers and Sequences") the authors give the same equations, without including the std, but with the cos and sin terms outside the radical. They call this the Box-Muller (umlaut over the u) method, by the way. They also give another way of generating a zero-mean, unit-variance Gaussian random variable:

$$Y = (\text{sum from } k=1 \text{ to } 12) (U(k) - 6.0)$$

where $U(k)$ is a uniformly distributed random number in the interval $[0,1]$.

-- Jon, KE3Z

16. Re: [HFSIG:86] Re: HF Modulation

From: "Johan Forrer
 Date: Wed, 9 Nov 1994 08:30:39 PST8PDT
 To: hfsig@tapr.org
 Message-ID: <7784F8A797F@frl.orst.edu>
 Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi folks,

Great contributions! Thanks.

```
>> WGN_real = std * sqrt (-2*ln(ran1)*cos(2*pi*ran2))
>> WGN_imag = .....Sin.....
>>
```

```
>> where: std is the std. dev. of GN,
>>         ran1, ran2 is uniformly distributed between 0,1

>Should the trig functions be inside the square root as shown? In the book
>referenced above, (in section 3.11, "Computer Generation of Random Numbers
>and Sequences") the authors give the same equations, without including the
>std, but with the cos and sin terms outside the radical. They call this the
>Box-Muller (umlaut over the u) method, by the way. They also give another
>way of generating a zero-mean, unit-variance Gaussian random variable:
>
>Y = (sum from k=1 to 12) (U(k) - 6.0)
>
>where U(k) is a uniformly distributed random number in the interval [0,1].
>
> -- Jon, KE3Z
```

Jon, you are right - the sqrt should only contain the "ln(ran)" term, the sin/cos should be outside.

Johan, KC7WW

17. Asst HFSIG Chair/Scribe Needed

From: Greg Jones <gjones@tenet.edu>
 Date: Wed, 9 Nov 1994 14:42:48 -0600 (CST)
 To: hfsig@tapr.org (HF SIG mailing)
 Message-ID: <199411092042.0AA15915@Kay-Abernathy.tenet.edu>
 Rendered body type: text/plain

Hi Folks.

Have really enjoyed reading the threads so far.

I have been speaking to Johan and we both feel that the HFSIG needs an asst chair/scribe. This position would handle writing up the quarterly report on the HF SIG for the TAPR PSR.

While all the threads are kept on file, having the overview report each quarter in the PSR makes what the HF SIG does a part of the running history of TAPR and digital communications.

Also, the part of the Asst Chair is to help Johan organize things as they need to be done regarding the SIG from time to time when he does not have time. As well, the ast chair would be responsible if anything happens to the chair in the future. Keeping continuity is very important and the asst chair position will help with that.

If you think you are interested, contact Johan or me.

Cheers - Greg

18. CCIR via GOPHER

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
 Date: Wed, 9 Nov 94 16:20:21 CST
 To: hfsig@tapr.org
 Message-ID: <9411092220.AA07033@eagle.aud.alcatel.com>
 Rendered body type: text/plain

For those who may be interested, you can get the complete listing of CCIR recommendations, and the text (with figures) of some of them electronically. The following gopher is available. I accessed it recently, and there's a wealth of information on line. This are the instructions I downloaded from the README file....

```
>
> You can access the ITU/TIES information via gopher:
>
>   - use your own gopher client and access to info.itu.ch on port 70
>   or
>   - telnet to gopher.itu.ch on port 2000
>
> Please send any message to helpdesk@itu.ch
>
>
>                                     The TIES group
>
```

A search indicates that the most current version of CCIR 549 is -1, and it is about sidetone on handset for Shipboard radiotelephone (there was a reference to this by one post recently). Not sure if I copied the number correctly or not! The other one, was 520-2, and -2 is the current version. It is on HF channel simulation. Unfortunately it is not available electronically.

- Tom, N5EG

19. Re: [HFSIG:89] CCIR via GOPHER

From: "Johan Forrer"
Date: Wed, 9 Nov 1994 16:04:07 PST8PDT
To: hfsig@tapr.org
Message-ID: <77FDE6D00EF@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

> For those who may be interested, you can get the complete
>listing of CCIR recommendations, and the text (with figures) of some of
>them electronically. The following gopher is available. I accessed it
>recently, and there's a wealth of information on line. This are the
>instructions I downloaded from the README file....
>
>>
>> You can access the ITU/TIES information via gopher:
>>
>> - use your own gopher client and access to info.itu.ch on port 70
>> or
>> - telnet to gopher.itu.ch on port 2000
>>
>> Please send any message to helpdesk@itu.ch
>>
>> The TIES group
>>
>
>
>A search indicates that the most current version of CCIR 549 is -1, and it
>is about sidetone on handset for Shipboard radiotelephone (there was a
>reference to this by one post recently). Not sure if I copied the number
>correctly or not! The other one, was 520-2, and -2 is the current version.
>It is on HF channel simulation. Unfortunately it is not available
>electronically.
>
>
> - Tom, N5EG

Hi Tom,

It would be helpful to access the ITU/CCIR sources electronically. I thought that it was only available to certain member organizations - but maybe not. I'm not sure about the nomenclature - the documents that I have in front of me are titled:

" Report 549-2, HF Ionospheric channel simulators", and
" Recommendation 520-1, Use of high frequency ionospheric simulators".

Is it possible that the recommendations and reports are different documents? Can anyone help clarify this?

I read the Ehrman paper - quite interesting and takes you a step closer to doing it: there they sampled the input signal at 51.6ksps, filtered it, decimated 6:1 before Hilbert-transforming it to produce real and imag. components for the tapped delay lines. A fairly understandable description follows on how each of the delayed paths are processed to introduce Raleigh fading and doppler. Also given are how broad-band and impulse noise are treated. On the practical side, seems like they had the randomized Gaussian number generator and complex exponential generator as table lookups. The question remains whether this could all be done using fixed point or whether it needs floating point. It would be real nice if it could all be done on the DSP hardware without too much involvement from the host (except for configuration and downloading). Would someone like to comment on that?

Johan, KC7WW

20. Ehrman paper

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Thu, 10 Nov 94 10:44:45 CST

To: hfsig@tapr.org
Message-ID: <9411101644.AA07689@eagle.aud.alcatel.com>
Rendered body type: text/plain

I got a copy of the Ehrman paper (IEEE ComTech Vol. 30, No. 8, August 1982), "Real-Time Software Simulation of the HF Radio Channel". This is a really good how-to paper. It clarifies many of the thoughts that were left somewhat vague by the Watterson paper. Highly recommended reading.

- Tom, N5EG

21. Re: [HFSIG:86] Re: HF Modulation

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Fri, 11 Nov 94 09:42:29 CST
To: hfsig@tapr.org
Message-ID: <9411111542.AA08376@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Should the trig functions be inside the square root as shown? In the book
> referenced above, (in section 3.11, "Computer Generation of Random Numbers
> and Sequences") the authors give the same equations, without including the
> std, but with the cos and sin terms outside the radical. They call this the
> Box-Muller (umlaut over the u) method, by the way. They also give another
> way of generating a zero-mean, unit-variance Gaussian random variable:
>
> $Y = (\text{sum from } k=1 \text{ to } 12) (U(k) - 6.0)$
>
> where $U(k)$ is a uniformly distributed random number in the interval $[0,1]$.
>
> -- Jon, KE3Z
>

I used EXCEL to generate 500 random numbers using Jon's formula, and plotted the histogram, both pdf (probability density) and cdf (cumulative density). Both the pdf and the cdf look very much like those for gaussian functions. The mean is zero (obvious by inspection) and the standard deviation is 1.0 (not so obvious by inspection?). Given the lack of needing a log() or ln() function, this formula may be easier to generate on a fixed-point DSP engine. Ehrman gives a simple formula for a random number generator if a 32-bit accumulator is available. Thanks for the info, Jon.

- Tom, N5EG

22. Bandwidth

From: chbrain@dircon.co.uk (Charles Brain)
Date: Sat, 12 Nov 1994 11:08:03 GMT
To: hfsig@tapr.org
Message-ID: <199411121108.AA04501@felix.dircon.co.uk>
Rendered body type: text/plain

I just thought you might like to know what the U.K licence says about bandwidth.

" The bandwidths of emmissions should be such as to ensure the most efficient utilisation of the spectrum; in general this requires that bandwidths be kept at the lowest values which technology and the nature of the service permit. Where bandwidth-expansion techniques are used, the minimum spectral power density consistent with efficient spectrum utilisation should be employed."

This I take to mean, is that there is nothing precluding the use of voice bandwidth data modems on H.F.

Just got my QEX interesting article Johan.

Regards Charles

| Charles Brain (G4GU0) |
| Chelmsford, Essex. |
| E-mail chbrain@dircon.co.uk |
| POTS +44 (0)1245 353221 |
FAX +44 (0)1245 275448

23. Channel simulator math

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Mon, 14 Nov 94 08:10:28 CST
To: hfsig@tapr.org
Message-ID: <9411141410.AA12712@eagle.aud.alcatel.com>
Rendered body type: text/plain

Johan:

I spent some time reviewing the Ehrman article, and this is my understanding of the math involved:

The input is real-valued, but a pseudo-imaginary stream is generated via a hilbert transform (wideband 90-degree phase shift). I found a suitable program for generating coefficients in Burrus & Parks, "Digital Filter Design" but do not have access to the reference [7, Rabiner] where the efficiency can be doubled by using an odd number of taps. Interestingly, an even number of taps generates a high-pass transform, an odd number of taps must generate a band-pass transform [B&P].

Two independent gaussian-distributed random variables modulate the I,Q components of the vector modulator for each tap. The random numbers are filtered by a low pass filter with appropriate time constant. I am guessing that a random number that is low pass filtered still has the same random distribution statistics as before filtering (except for frequency of course), this appears to be Ehrman's assumption.

The Doppler shift is done by a vector modulator with the variable being a constant-rotating phase shift, but with no random component. The rotating phase generates the + or - doppler frequency offset. The gaussian frequency doppler spread is caused by the first modulator (in the previous paragraph) since it had to contribute random phase changes.

Does this agree with your interpretation?

Johan, I very much appreciate the references to the various articles, as they speed up considerably finding and understanding the material! Trying to find all the relevant articles is slow and laborious, and that's one of the good things about the newsgroup - getting those important references found. I did acquire a copy of both CCIR reports/recommendations you suggested, but have not read them yet.

- Tom, N5EG

24. Re: [HFSIG:94] Channel simulator math

From: "Johan Forrer"
Date: Mon, 14 Nov 1994 08:32:14 PST8PDT
To: hfsig@tapr.org
Message-ID: <7F0595A4D2D@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

>Johan:

>

> I spent some time reviewing the Ehrman article, and this is my
>understanding of the math involved:

>

> The input is real-valued, but a pseudo-imaginary stream is generated
>via a hilbert transform (wideband 90-degree phase shift). I found a
>suitable program for generating coefficients in Burrus & Parks, "Digital
>Filter Design" but do not have access to the reference [7, Rabiner] where
>the efficiency can be doubled by using an odd number of taps.
>Interestingly, an even number of taps generates a high-pass transform, an
>odd number of taps must generate a band-pass transform [B&P].

Yes, an analytical signal is formed, though of course you may also achieve similar results by down-converting in frequency domain at the input, and a similar up-conversion at the output, however, using the Hilbert pair saves a few computations. One thing that are implied in most of the papers so far, is that, if you do use the Hilbert transformer, its delay must be compensated by a similar delay in the "I" branch. A neat description of this is given in: "Theory and Implementation of a Slitband Modem Using the TMS32010" by George Troullinos et al. SPRA013. If you ask nicely, the folks at TI's Literature Distribution will give it away for free. I actually have based my simulator on the filters that are given in this booklet. If you look at the coefficients that they provide in the above article, you will see the effect of the odd number of taps, i.e. half the coefficients are zero. There is, however, some concern of spurious harmonics due to imperfections in the transform that should ideally be removed by LPF. I'm not too sure how important all this would be in real applications.

> Two independent gaussian-distributed random variables modulate the

>I,Q components of the vector modulator for each tap. The random numbers
>are filtered by a low pass filter with appropriate time constant. I am
>guessing that a random number that is low pass filtered still has the same
>random distribution statistics as before filtering (except for frequency
>of course), this appears to be Ehrman's assumption.
>
> The Doppler shift is done by a vector modulator with the variable
>being a constant-rotating phase shift, but with no random component. The
>rotating phase generates the + or - doppler frequency offset. The gaussian
>frequency doppler spread is caused by the first modulator (in the previous
>paragraph) since it had to contribute random phase changes.
>
> Does this agree with your interpretation?

Yes, basically this is how I understand it too, and also from the plots that I have generated, looks feasible. Multiplication is distributive of course and you may perform the multiplications in any order. What I do is to first generate the complex Gaussian signal, LPF, then mix the doppler signal and last, mix the complex signal from the input delay line. You may find however, that the sample rate at the Gaussian/doppler pair are at much lower rate than the input sampling, i.e. around the hundred SPS range (due to the very low frequencies generated at that point) vs. the input signal needs to be sampled at least twice 3 KSPS (which incidentally, I choose 9600 SPS). This means that you will need an interpolating filter at the gain tap functions else the signals there will be very ragged steps.

This brings me to another issue that the Ehrman paper is very loose on. If you are creating three signals that need to be combined, you need to ensure that they have equal power - how do you go about that? Also if the Gaussian generator has to be LPF'd, it will change its power, so you need to compensate for that when you choose your variance when generating the Gaussian signal.

> Johan, I very much appreciate the references to the various articles,
>as they speed up considerably finding and understanding the material!
>Trying to find all the relevant articles is slow and laborious, and that's
>one of the good things about the newsgroup - getting those important
>references found. I did acquire a copy of both CCIR
>reports/recommendations you suggested, but have not read them yet.
>
> - Tom, N5EG

Ditto - goes without saying that I am grateful for all the participation. I do hope that you will consider writing all of this up some time in a article?

73's

Johan, KC7WW

25. Re: [HFSIG:93] Bandwidth

From: "Johan Forrer
Date: Mon, 14 Nov 1994 08:38:55 PST8PDT
To: hfsig@tapr.org
Message-ID: <7F0763671E2@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Charles,

>I just thought you might like to know what the U.K licence says about
>bandwidth.

>" The bandwidths of emmissions should be such as to ensure the most
>efficient utilisation of the spectrum; in general this requires that
>bandwidths be kept at the lowest values which technology and the nature of
>the service permit. Where bandwidth-expansion techniques are used, the
>minimum spectral power density consistent with efficient spectrum
>utilisation should be employed."

>
>This I take to mean, is that there is nothing precluding the use of voice
>bandwidth data modems on H.F.

28. Re: Channel simulator math

From: Eric E Silbaugh <esilbaug@afit.af.mil>
Date: Mon, 14 Nov 1994 20:22:57 -0500
To: hfsig@tapr.org
Cc: esilbaug@afit.af.mil
Message-ID: <199411150122.AA02585@stealth.afit.af.mil>
Rendered body type: text/plain

Tom ,

If you want more info on Hilbert transforms you might try looking for "Discrete-time Signal Processing" by Oppenheim and Schaffer. It's one of the 'classic' DSP texts and should be in most university libraries. Chapter 10 is devoted entirely to Hilbert transforms.

Johan's comments on delay and coefficients are exactly correct. Now for more gory details. You probably also noticed that the coefficients of the Hilbert transform have odd symmetry. In other words, $X[n] = -X[-n]$. These coefficients have the same magnitude but opposite sign. Now, combine this with the fact that half the coefficients are identically zero for Hilbert transforms with an odd number of taps. (Note that Hilbert transforms with an even number of taps have non-zero coefficients for all taps)

Because of these nice properties we don't have to compute the response for half the taps. We can also take the difference of the input data corresponding to taps with equal magnitudes and multiply this difference once by the coefficient. These properties reduce the number of multiplications by a factor of 4 and halve the number of additions. Pretty good, and we haven't even broken a sweat!

Another nice property is that these Hilbert transforms have linear phase because of the symmetric coefficients. You get an *exact* 90 degree phase shift, plus a constant delay. The problem is that the magnitude response has ripples (Yes, like the potato(e) chip).

The Ehrman paper mentions having an error of less than -50 dB (I presume they mean ripple error) in their Hilbert transform. In my experience this would mean a filter with ~40-60 taps, depending on the required bandwidth. The number of taps required increases quickly if you need accurate response near the band edges (0 Hz and Nyquist frequency).

Well, now that I've completely bored you...later.

Eric

Eric E. Silbaugh
esilbaugh@afit.af.mil

All standard, non-standard, and
MIL-STD disclaimers apply

29. Documentation

From: Eric Rosenberg <ericr@lan.vita.org>
Date: Tue, 15 Nov 1994 12:20:47 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.3.89.9411151206.A16114-0100000@lan.vita.org>
Rendered body type: text/plain

This is a bit off the technical track, but I am looking for a clear, concise publication that explains hf communications to the lay person. CODAN, the Australian manufacturer, used to put out such a book, but alas, I need something more vendor generic.

At VITA, we field HF packet radio systems in developing countries, and explaining why things do or don't work in their language is the toughest job of all!

Thanks --
Eric

--
Eric Rosenberg
Volunteers In Technical Assistance voice: +703-276-1800
1600 Wilson Blvd., Suite 500 fax: +703-243-1865
Arlington, VA 22209 USA ericr@vita.org

30. Ideas for HF modulation and coding

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Nov 1994 16:22:12 -0800
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199411160022.QAA01412@servo.qualcomm.com>
Rendered body type: text/plain

After the last Dayton Hamvention, W3IWI and I drove to Baltimore and we discussed at length some ideas I had for a "moonbounce modem". We eventually came to the conclusion that if my scheme worked via the moon, it should work even better over HF.

First, some background.

The basic problem is that you're operating over a dispersive, fading channel. Received phase is essentially random, so coherence is hard to achieve. But it turns out that you can buy back much of the loss of noncoherent demodulation -- at the cost of increased bandwidth -- with m-ary orthogonal signalling.

"Orthogonal" simply means that the signal states are independent of each other. You can build a detector that responds to one signal state while remaining completely unaffected (in theory) by the other states. Plain mark/space FSK is 2-ary orthogonal signalling, assuming the shift is large enough for the signalling rate. You can detect FSK with two properly tuned filters, and (again in theory) the mark filter does not respond at all to a space signal, and vice versa.

Ordinary BPSK is not orthogonal, it's antipodal. One state is the exact opposite of the other, so instead of being independent, they "fight" each other much like a tug-of-war contest. You use a single detector and compare the result to a threshold. Antipodal modulation is 3dB better than orthogonal modulation but it requires phase

coherence, which we don't have on HF.

So we're stuck with orthogonal signalling. But instead of being limited to only two antipodal signal points, you can have as many orthogonal signal points as you like. In m-ary orthogonal signalling, you have m possible channel states, where m is usually a power of 2. E.g., 4-tone FSK is a 4-ary orthogonal modulation scheme. Each channel symbol carries 2 bits of information in the $2^2 = 4$ possible states, assuming only one of the four tones is on at any time (otherwise you'd just have several 2-ary systems in parallel.)

It turns out that as m approaches infinity, overall performance asymptotically reaches the theoretical Shannon limit of -1.6dB Eb/N0, even with noncoherent demodulation! This has been known for a long time, but unfortunately the improvement with m is too slow to be impractical.

Given that, here's my idea. Take a reasonable channel bandwidth, say 2 Khz since that will fit through most SSB filters. Divide this 2 Khz into 256 frequency bins, each 7.8125 Hz wide. In order to keep the bins independent and orthogonal, our signalling rate will be limited to no more than 7.8125 symbols per second. But we want a slow symbol rate anyway to minimize intersymbol interference caused by ionospheric multipath. Also, synchronization is easier at a low symbol rate; it may even be possible to use GPS with approximate propagation delay figures to synchronize "open loop". Anyway, with that many bins we could send $\log_2(256) = 8$ bits per symbol, or 62.5 bits/sec.

Although this scheme by itself is orthogonal, it isn't particularly resistant to selective fading or narrowband interference. To improve this I propose another layer of orthogonal coding using Walsh functions (rows of a Hadamard matrix).

A Hadamard matrix is easily built up recursively. You start with a 1x1 matrix containing the single entry "1". Then you double each dimension of the matrix by copying it to the right and below unchanged, and diagonally to the lower right by inverting each bit. Here's the recurrence relation defined using the limited graphics of ASCII:

$$M(i+1) = \begin{matrix} & M(i) & M(i) \\ M(i) & & \sim M(i) \end{matrix}$$

You can keep repeating this process until you have a matrix of the

desired size. Here are the first few Hadamard matrices:

```
1
11
10

1111
1010
1100
1001

11111111
10101010
11001100
10011001
11110000
10100101
11000011
10010110
```

and so forth.

Now notice that if you take any two rows at random from a Hadamard matrix and XOR them bitwise, you'll always get an equal number of 1's and 0's. It's easy to see how this follows from the recursive definition given above.

This property means that the rows (and the columns, for that matter) of a Hademard matrix form a set of mutually orthogonal functions. These are called Walsh functions, and have been around for a long time. (As an aside, one clever use in the early days of telephony was to set up wire-crossing patterns for parallel open-wire transmission lines in order to minimize inductive crosstalk.)

Now let's say that instead of just turning on one tone (out of 256) at a time for each transmitted symbol, we take the 8-bit symbol and index the corresponding row from a 256x256 Hadamard matrix. The 256 bits in that row then control whether or not the tone is turned in each of 256 frequency bins. (The amplitude of each tone is reduced to keep the transmitter power the same.)

Each symbol still carries only 8 bits, so the energy per bit is the same in this scheme as with the simple 1-tone-out-of-256 scheme. Moreover, for every Walsh codeword except the all-1s case, exactly 50% of the bins are on and 50% are off, so the transmitter power is constant. (The all 1's case could be avoided with some work if necessary.)

So performance with and without Walsh coding should be exactly the same over a nonfading gaussian noise channel. But HF has both non-gaussian (e.g., narrowband) interference and fading, often of the frequency-selective kind. Walsh encoding should help by spreading each symbol's energy out more evenly over the frequency channel.

You could demodulate this signal by first taking a FFT of the raw input samples to determine the total received energy in each frequency bin. Then you take the FHT (Fast Hademard Transform, analogous to the FFT) to determine which of the 256 Walsh functions was mostly likely to have been the one that was sent.

Note that the first column in each of the Hademard matrices is a 1, which is rather redundant. You could omit it and save a frequency bin (which is pretty minor). This is called a "simplex" code, as opposed to a straight orthogonal code. But you could do much better by augmenting the matrix with each bit inverted, doubling the number of code words and adding a bit to each symbol. For example, if we take the 4x4 Hademard matrix

```
1111
1010
1100
1001
```

and append to it its bit-inverse, we get

```
1111
1010
1100
```

1001
0000
0101
0011
0110

The 8 rows are now either mutually orthogonal (equal number of bit agreements and disagreements) or antipodal (all bits disagree).

This is called a "biorthogonal" code, and in fact it's one of the earliest error correcting codes used in spacecraft. Mariner 9 used one (64x32, I think) at Mars in the late 1960s, before the big switch to convolutional coding.

The only reason to prefer a simplex or orthogonal code set over a biorthogonal code is when your channel has a polarity ambiguity (e.g., BPSK). You could easily determine polarity with an orthogonal or simplex code, but not with a biorthogonal code. But we don't have that problem here, since each "bin" is basically an on-off keyed channel without any polarity ambiguity.

If we built a 128x128 Hademard matrix and turned it into a biorthogonal code we could send 8-bit symbols with only 128 frequency bins. This would let us double our symbol rate to 15.625/sec, or our raw bit rate to 125 bits/sec. Or we could halve the transmitted bandwidth and keep the earlier data rate (more on bandwidth issues later).

Although this scheme should be very robust, there will be bursts of QRM and deep flat fades too great for it to handle. We could deal with this by adding another layer of FEC. A natural choice would be a Reed-Solomon block code over GF(2⁸). This code operates on 8-bit symbols that match the basic channel symbol. Paul, N9FZX has already implemented a Reed-Solomon decoder for 8-bit symbols and has made it freely available over the net. Although the natural block size for a RS coder over GF(256) is a 255-byte block, you can use a smaller blocksize if you like by just agreeing between sender and receiver that the unused symbols are 0s and not sending them. On the other hand, larger blocksizes are more robust for the same amount of redundancy.

A more complicated scheme that might work better is convolutional encoding with soft decision decoding, either Fano or Viterbi. The convolutional decoding itself is not a big problem (I've already implemented and tested both decoders) but what makes it complicated is the need to modify the Walsh function demodulator to provide soft samples (instead of "hard" 1s and 0s) for each bit of each demodulated symbol. There are ways to do this, but they involve a fair amount of effort. I need to look into this.

I also need to look into the optimality of the biorthogonal scheme. For maximum efficiency you like to have a channel code where every possible error event is of equal probability, but the biorthogonal code doesn't do this. (The probability of one codeword being turned into its antipodal complement is obviously much less than the probability of it being turned into another codeword that is merely orthogonal to it, since the hamming distance between antipodal codewords is twice that of the distance between two orthogonal codewords.) If you can't keep the codewords at equal distance, it might help to take them into account in your upper layer FEC design. Or it might not make a big difference, I don't know.

I'm aware that mixing a large number of discrete tones would increase the variance of the instantaneous transmitter power. My intuition and the central limit theorem tells me that the instantaneous envelope power will end up looking very Gaussian. But Shannon says that efficient modulation schemes *have* to look Gaussian, and it may turn out that the increased coding gain of this scheme more than makes up for any increased amplifier losses over a constant-envelope signal.

So in summary, my scheme would take a block of 8-bit data bytes, add an appropriate number of Reed-Solomon FEC symbols, encode each symbol into a (128,8) biorthogonal channel code, and then use each of the 128 channel bits to gate a set of 128 discrete channel tones.

Some remarks on bandwidth. The bandwidth efficiency of this scheme may at first glance appear to be unacceptably low, especially for HF. 125 bits/sec in 2 KHz is only 0.0625 bits/sec/Hz. But the more I learn about information theory both from the textbooks and from my work here

at Qualcomm, the more I really understand that it's counterproductive to limit signal bandwidth. In the real world of a shared band, power efficiency and interference resistance are FAR more important, and these two properties are possible ONLY with extra bandwidth.

This particular modulation scheme should be incredibly robust against both noise and QRM. It ought to be easily decodable well below the operator's ability to even hear it. Because of its spread spectrum-like properties, strong inband QRM can be notched out fairly easily. Since the energy distribution of the signal is fairly flat, after the FFT you just look for any frequency bins with energies way above average and suppress them before the FHT step. The only limit on the ability to suppress narrowband QRM (a CW carrier, say) then becomes the dynamic range of the receiver and A/D converter.

So, what do people think? This stuff represents a radical departure from the usual approach of trying to minimize congestion by cramming as much as possible into a narrow HF channel. Because of its wide bandwidth, we'd need STAs to test my scheme on the HF bands, and a rule change for general use. But the FCC's bandwidth rules are based on intuition that happens to be completely wrong. John Costas said as much in his classic paper "Poisson, Shannon and the Radio Amateur", published 35 years ago, but now we actually have the technology to pursue his suggestions!

Phil

31. Re: Ideas for HF modulation and coding

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Nov 1994 16:26:33 -0800
To: karn@qualcomm.com
Cc: hfsg@tapr.org
Message-ID: <199411160026.QAA01424@servo.qualcomm.com>
Rendered body type: text/plain

>time, but unfortunately the improvement with m is too slow to be
>impractical.

Uh, minor typo. That should be "too slow to be PRACTICAL by itself".

Phil

32. An Introduction

From: Eric Rosenberg <ericr@lan.vita.org>
Date: Wed, 16 Nov 1994 10:21:23 -0500 (EST)
To: hfsg@tapr.org
Message-ID: <Pine.3.89.9411161028.A24081-0100000@lan.vita.org>
Rendered body type: text/plain

In order to bring some levity to the heavy conversations on modulation schemes, I think it's time I add my two cents by way of introducing myself and expressing my interest in HF digital radio.

My name is Eric Rosenberg, my callsign is WD3Q, and I live in Washington, DC. I've been licensed for 20+ years, and have lived and operated from NY, CA, SD, OH and PA before moving to DC in 1988.

In addition, for the past three years my job has kept me overseas for long periods of time (too much, my wife says!), so I've managed to get licenses and operate from 9L, EI, F, G, J2, VK, YJ, and ZL (and listened from places where licenses were not available: YB and 9V).

I work for a non-government development group called VITA: Volunteers in Technical Assistance. Many of you have heard of us through our satellite activities...we were the prime movers behind the Digital Communications Experiment on UO-11 and are the license holders and primary operators of the non-amateur activity on the UO-14 satellite. In addition to the satellite work, we have also been involved in the design and implementation of terrestrial radio systems, mostly in Africa and Asia.

To date we've installed systems in Chad, Sudan (2 systems), Philippines, Madagascar, and have assisted in the installation of systems in Lesotho, Sierra Leone, to name a few. The systems started out using AX.25, but have moved to 1200 bps PSK and more rrecently PacTOR. The distances between stations generally range from 200 - 1000km, with the average being 350 km. As the MUFs start moving down, we're now revisiting what we're putting out in the field.

Our issues have rarely been technical...it is less important what modulation scheme is used than ability of that modem to be installed on a wide range of radios, from the Yaesu FT-747 (probably the single most common radio found in developing countries) to the more expensive commercial units made by Codan, Transworld, Harris, et al. It seems that each group we deal with has a smattering of radios, no two from the same manufacturer or age!

The most important issue is usability. Can the average, not terribly technically-minded person use this system effectively without having to become a trained radio operator.

In the idealized world, we would like to see a system that incorporated some form of dynamic routing and ALE (Transworld's Transcall would suffice), user friendly software in the language of the user, and modems using modulation schemes that would interface to the wide range of radios we encounter. We've also done some work incorporating embedded controllers with the modems (this was done for a hands off, automated satellite network in Indonesia), and that looks like the real way to go.

There's more, but these are the basics.

To get to where we're going, VITA has just started to develop some unique applications, and hope to put up a multi-station experimental network to test it all out. We'll solicit you for help as it develops.

One final note...the "Volunteers" in our name refers to you. We've had great participation from the amateur radio community in developing and fielding satellite and terrestrial systems. Rick Whiting, W0TN, did our first HF demo in Ethiopia in the mid 1980's, and we've had great help from KD6KQ, N3EOY, EI9GL, W2JUP, HB9AQZ, SM0TER, N6ARE and others. Norm Sternberg and now Rick Whiting and Dave Henderson (N3EOY) teach a class given yearly on low cost digital radio systems for the US Telecommunications Training Institute, a government-private partnership program that brings communications professionals from developing countries to the US for advanced training.

I've been fascinated by the more esoteric discussions here. Just don't lose track of the users, please!

Eric

--

Eric Rosenberg
Volunteers In Technical Assistance
1600 Wilson Blvd., Suite 500
Arlington, VA 22209 USA

WD3Q, J20BY, YJ0AER, VK2GYA et al
voice: +703-276-1800
fax: +703-243-1865
ericr@vita.org

33. Re: [HFSIG:95] Re: Channel simulator math

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Wed, 16 Nov 94 15:54:15 CST
To: hfsig@tapr.org
Message-ID: <9411162154.AA01548@eagle.aud.alcatel.com>
Rendered body type: text/plain

Johan writes:

>
> Yes, basically this is how I understand it too, and also from the plots that
> I have generated, looks feasible. Multiplication is distributive of course
> and you may perform the multiplications in any order. What I do is
> to first generate the complex Gaussian signal, LPF, then mix the doppler
> signal and last, mix the complex signal from the input delay line. You may
> find however, that the sample rate at the Gaussian/doppler pair are at
> much lower rate than the input sampling, i.e. around the hundred SPS
> range (due to the very low frequencies generated at that point) vs. the
> input signal needs to be sampled at least twice 3 KSPS (which incidentally,
> I choose 9600 SPS). This means that you will need an interpolating filter at
> the gain tap functions else the signals there will be very ragged steps.
>
> This brings me to another issue that the Ehrman paper is very loose on. If
> you are creating three signals that need to be combined, you need to ensure
> that they have equal power - how do you go about that? Also if the Gaussian
> generator has to be LPF'd, it will change its power, so you need to
> compensate for that when you choose your variance when generating the
> Gaussian signal.
>

It would seem that we could perform the interpolation after the

multiplication and summation of all the components. As I recall, interpolation is really low-pass filtering, and as convolution (FIR filter) is associative (like multiplication), we should be able to just low-pass-filter the final output signal, assuming that no spurious mixing products are generated anywhere downstream of them complex Gaussian stream. Is this correct, and would it save a significant amount of processing?

Also, it does not seem intuitive that we need to actually carefully match the received power in the three different components, since I see no reason why all three propagation components would actually have exactly the same efficiency in the real world. One of the CCIR documents recommends testing with the three components set to different relative power and to the same relative power over about a 40 db range.

Thanks for the informative comments from Johan and Eric. Hopefully this is all getting clear now.

- Tom

34. Re: [HFSIG:101] Ideas for HF modulation and coding

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Wed, 16 Nov 94 16:05:45 CST
To: hfsig@tapr.org
Message-ID: <9411162205.AA01552@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Now let's say that instead of just turning on one tone (out of 256) at
> a time for each transmitted symbol, we take the 8-bit symbol and index
> the corresponding row from a 256x256 Hadamard matrix. The 256 bits in
> that row then control whether or not the tone is turned in each of 256
> frequency bins. (The amplitude of each tone is reduced to keep the
> transmitter power the same.)
>
> Each symbol still carries only 8 bits, so the energy per bit is the
> same in this scheme as with the simple 1-tone-out-of-256
> scheme. Moreover, for every Walsh codeword except the all-1s case,
> exactly 50% of the bins are on and 50% are off, so the transmitter
> power is constant. (The all 1's case could be avoided with some work
> if necessary.)

...

> Phil
>

Phil: a question: under this scheme, I agree that the power is constant, however it would seem that the peak amplitude could be very high. Would this require a great deal of linearity in the transmitter and receiver?

- Tom, N5EG

35. Re: [HFSIG:105] Re: Ideas for HF modulation and coding

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 16 Nov 1994 19:44:47 -0800
To: hfsig@tapr.org
Message-ID: <199411170344.TAA08291@servo.qualcomm.com>
Rendered body type: text/plain

> Phil: a question: under this scheme, I agree that the power is
> constant, however it would seem that the peak amplitude could be very
> high. Would this require a great deal of linearity in the transmitter
> and receiver?

Yes, it would. I address this particular point later on. I assert (without proof) that the coding gains would make up for any losses due to having to back off your amplifier to keep it linear.

Phil

36. Re: [HFSIG:100] Documentation

From: rs@ife.ee.ethz.ch (Rolf Sommerhalder)
Date: Thu, 17 Nov 94 9:13:19 MET
To: hfsig@tapr.org
Message-ID: <9411170813.AA00226@ife.ee.ethz.ch>
Rendered body type: text/plain

> At VITA, we field HF packet radio systems in developing countries, and
> explaining why things do or don't work in their language is the toughest
> job of all!

Hi Eric

Did you see the HF product catalog of SGC Corp. ? They have some
general description of the art of HF radio comms in there which
might be the level you are looking for. I can look up that
company's address if you like to contact them.

vy 73,
Rolf
(Swiss Disaster Relief, Telecom Group)

P.S.
We are enjoying VITA's WWW server to get the latest DHA reports...

--

--- * * * ---

Rolf Sommerhalder	
Swiss Federal Institute of Technology (ETH)	home address:
Electronics Laboratory ETZ H60.1	c/o Rindlisbacher
Gloriastrasse 35	Wermatswilerstrasse 96
8092 Zurich / Switzerland	8610 Uster / Switzerland
Voice +41 1 632 76 40 (or 632 51 53)	Voice +41 1 941 59 28
Fax +41 1 632 12 10	AX.25 HB9CWP @ HB9OS
E-Mail rolf.sommerhalder@ife.ee.ethz.ch	

37. HF Simulator test results

From: "Johan Forrer"
Date: Fri, 18 Nov 1994 09:10:42 PST8PDT
To: HFSIG@tapr.org
Message-ID: <851008E47FD@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi All,

I have made a small test file containing test results from my efforts
on the HF simulator. It is available for downloading via anonymous FTP from:

ucs.orst.edu /tmp/HFSIM.ZIP

This test was generated using a C program and simulating CCIR "GOOD",
"MODERATE", "POOR" and "Flat fading" conditions. I still have to look more
closely at intermediate results to check whether the magnitudes are OK, but
results look like what is expected (I think?).

I would appreciate your feedback on these results.

The next step that I'm working on, is to implement the Hilbert transform,
tapped delay lines, and complex mixers on the sound card. Since the
Gaussian noise & doppler gain tap functions operates at a very low sample
rate, I'm planning to let the PC generate these, and transfer the results
over the ISA bus to the sound card for real time mixing. Perhaps I'll be
able to listen to some real time simulations this weekend.

73's

Johan

38. Re: [HFSIG:108] HF Simulator test results

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Fri, 18 Nov 94 11:39:21 CST
To: hfsig@tapr.org
Message-ID: <9411181739.AA03799@eagle.aud.alcatel.com>
Rendered body type: text/plain

> Hi All,
>
> I have made a small test file containing test results from my efforts
> on the HF simulator. It is available for downloading via anonymous FTP from:
>
> ucs.orst.edu /tmp/HFSIM.ZIP
>
> This test was generated using a C program and simulating CCIR "GOOD",

> "MODERATE", "POOR" and "Flat fading" conditions. I still have to look more
> closely at intermediate results to check whether the magnitudes are OK, but
> results look like what is expected (I think?).
>
> I would appreciate your feedback on these results.
> 73's
>
> Johan
>

Johan: congratulations on making so much progress on the simulator.
You are way ahead of the rest of us. In looking through the data, I do not
understand what the data represents. Is the numeric value a one-second
average of the amplitude from summing all the components?

- Tom, N5EG

39. Re: [HFSIG:101] Ideas for HF modulation and coding

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Fri, 18 Nov 1994 13:45:54 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941118132302.1911W-100000@daffyduck>
Rendered body type: text/plain

On Tue, 15 Nov 1994, Phil Karn wrote:

>
> Some remarks on bandwidth. The bandwidth efficiency of this scheme may
> at first glance appear to be unacceptably low, especially for HF. 125
> bits/sec in 2 KHz is only 0.0625 bits/sec/Hz. But the more I learn
> about information theory both from the textbooks and from my work here
> at Qualcomm, the more I really understand that it's counterproductive
> to limit signal bandwidth. In the real world of a shared band, power
> efficiency and interference resistance are FAR more important, and
> these two properties are possible ONLY with extra bandwidth.
>

Hmmm...

My point of reference is the work I've been doing here at Motorola. We
build RF modems which operate at 19.2 kbps over 25 kHz channels for an
efficiency of 0.768 bits/sec/Hz. The modulation scheme is 4-tone FSK with
Trellis coding and forward error correction. Of course, the Trellis coding
is overhead which reduces the effective bit rate. Does the larger
bandwidth that we have available allow us greater efficiency than that of
your proposed scheme? Do we approach some sort of limit (i.e. Shannon's)
as the system becomes more spread-spectrum like? BTW, our products operate
at VHF so we aren't too concerned about dispersion. However, we do have to
deal with fading as they were originally designed for mobile applications,
although they're finding their way into portable applications these days.

>
> So, what do people think? This stuff represents a radical departure
> from the usual approach of trying to minimize congestion by cramming
> as much as possible into a narrow HF channel. Because of its wide
> bandwidth, we'd need STAs to test my scheme on the HF bands, and a
> rule change for general use. But the FCC's bandwidth rules are based
> on intuition that happens to be completely wrong. John Costas said as
> much in his classic paper "Poisson, Shannon and the Radio Amateur",
> published 35 years ago, but now we actually have the technology to
> pursue his suggestions!
>

I'm trying to hoist it all in! I'm somewhat of a newbie to HF data so I
don't have too many preconceptions. If a wider channel allows greater
efficiency and better immunity to QRM then it's hard to argue against
this on strictly technical grounds. Of course there are politics & ;)

I haven't played with the sound card much yet. Is it fast enough to
perform the transforms that you mention? Maybe benchmarking these
algorithms would be a good place for me to get started? What do you say
Johan?

Hugh

40. Bibliography

From: "Rick Whiting" <Rick_Whiting@ATK.COM>
Date: 18 Nov 1994 17:40:02 -0600

Johan, KC7WW

forrerj@ucs.orst.edu
(I also am on CI\$ if you prefer that route).

43. HF modulation

From: "Johan Forrer"
Date: Tue, 22 Nov 1994 11:13:53 PST8PDT
To: HFSIG@tapr.org
Message-ID: <8B310D736EC@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi,

After just briefly browsing through Phil's post - my \$0.02.

It is true that for practical reasons, non-coherent techniques makes a lot of sense. I just wonder though, whether we should totally disregard the possibility of extracting phase information with some reliability over HF links - this factor will play a key role of how we should trade off spectral efficiency and bandwidth efficiency. My question is: do we know enough to say yes or no at this point in time? I suspect we don't.

If we judge from the experiences with of commercial MIL-STD-188 HF modems, we see that they do implement some of the things that Phil mentioned, i.e. orthogonal signalling in the form of 16 or 39 parallel carriers, however, each being phase modulated. I suspect that perhaps some tradeoff is possible and that is, IMHO, where we can made our contributions.

Coding theory could do wonders when done right and perhaps should be part of the modulation as Phil suggest, however, I hope we do justice to modulation basics. I hope the simulator would help in providing some clues.

If I had my druthers, I would have opted for SS, but that is perhaps looking too far into the future. Or is it?

73's

Johan, KC7WW

P.S. pse e-mail me at: forrerj@ucs.orst.edu

44. [HFSIG] HF Channel Simulator

From: BobH@eworld.com
Date: Sat, 26 Nov 94 04:06:48 PST
To: hfsig@tapr.org
Message-ID: <9411260406.tn15861@eworld.com>
Rendered body type: text/plain

H.F. Channel Simulator Validation
Or
Making Sure The Medium Is The Message
And Not The Massage *(very droll, no?)

*With apologies to Mr. McLuhan who I don't even pretend to understand.

I'm new to the sig so I hope this topic hasn't been covered, or even worse, is obvious.
In reading over the CCIR, it is clear that the statistics of the fading filter output are key to quality and appropriateness of the simulation. Other functions such as Doppler offset are more straight forward and easily measured (SNR calibration might also be a problem however).
IMHO a couple of items must be verified to show compliance or surely any simulated performance of waveform, coding or protocol will quickly degenerated into a challenge to the simulator's channel transfer function.
> From the CCIR "Two independent complex Gaussian ergodic random processes, each with zero mean values and independent real and imaginary components with equal r.m.s. values that produce Rayleigh fading..." must be generated. Any simulator validation must show the amplitude and phase of the complex fading envelope, for each path, to be Gaussian. This would include statistical tests for independence (to aid diversity schemes), mean, variance and a probability distribution histogram. Also, a simulator controller should provide the fade statistics to document any given test run.
> Not independent but different, again from the CCIR "Each tap-gain function has a spectrum...that in general consists of the sum of two magnetonic components, each of which is a Gaussian function of frequency" (the 2 sigma points are given as a parameter) and later "The importance of the shape of

the tap-gain spectrum should be noted". This "double-sided smear bandwidth" should be measured in frequency and shown to have zero mean and unit standard deviation of accuracy over 2 or 3 sigma.

Also, in my personal wish list for a channel simulator: the CCIR suggests a specular (non-fading) component for a ground wave which we encounter much in MARS work. Also, a radio filter for those insisting on practicality.

Bob, wm6h
bobh@eworld.com

December 1994

Original source file: hfsig.9412

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1994-12/TAPR-HF_1994-12.txt

1. HF Channel simulator

From: "Johan Forrer
Date: Tue, 6 Dec 1994 09:58:39 PST8PDT
To: HFSIG@tapr.org
Message-ID: <A01D8F158A9@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

I have received some extra copies of the CCIR report 549-2
"HF IONOSPHERIC CHANNEL SIMULATORS" from Barry. Anyone still in need for a
copy please e-mail me. Thanks Barry!

The simulator is coming along quite nicely - I have part implemented on the
sound card DSP and the rest on the 486. The "flat fading" model
is working in a fashion - the main problem presently is to make the 486 keep
up with the sound card. I found that I had to implement quite a bit of the
simulation work using table lookup methods instead of calls to the 486's FPU
transcendental functions - just too slow. I'll share with you soon as that I
have something that works reasonably well.

73's

--Johan, KC7WW

2. Re: [HFSIG:116] HF Channel simulator

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Tue, 6 Dec 94 12:41:37 -0600
To: hfsig@tapr.org
Message-ID: <9412061841.AA29948@sacdm10.kelly.af.mil>
Rendered body type: text/plain

In your message of 6 Dec 1994 at 1204 CST, you write:

> Hi all,
>
> I have received some extra copies of the CCIR report 549-2
> "HF IONOSPHERIC CHANNEL SIMULATORS" from Barry. Anyone still in need for a
> copy please e-mail me. Thanks Barry!
>
> The simulator is coming along quite nicely - I have part implemented on the
> sound card DSP and the rest on the 486. The "flat fading" model
> is working in a fashion - the main problem presently is to make the 486 keep
> up with the sound card. I found that I had to implement quite a bit of the
> simulation work using table lookup methods instead of calls to the 486's FPU
> transcendental functions - just too slow. I'll share with you soon as that I
> have something that works reasonably well.
>
>
> 73's
>
> --Johan, KC7WW
>

I too received a copy of the CCIR...thanks a whole bunch Barry. Now I
will have to read it and try to understand it. Where's my old K&E
slipstick?

I too can make copies of the CCIR and will be happy to mail them...
just E-Mail me at k5yfw@sat.ampr.org.

FOR BARRY, I'd be very interested in getting the rest of REPORT 111,
INFLUENCE ON LONG-DISTANCE HF COMMUNICATIONS...that starts on the
last page of the CCIR you sent. I'll send postage.

In other news, General xxxx (he doesn't want to be identified) of the
Army, Army/MARS program wants *all* MARS traffic to start going by
high speed data...HF/VHF/UHF, etc. I think he's sold on PACTOR. Ok
guys#...time to shine and I *KNOW* you can do it.

Keep those E-Mail messages flowing...this is the most interesting
mailgroup I subscribe to...and probably the most important.

I'm anxiously awaiting for my copy of the CD "Seek You". See the review in the Jan '95 QST.

73,

Walt (dba k5yfw)

gender neutral from the NY "youse guys".

3. HF Channel simulator

From: "Johan Forrer"
Date: Mon, 12 Dec 1994 08:54:21 PST8PDT
To: HFSIG@tapr.org
Message-ID: <A90CB2538A1@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Greetings,

I am looking for volunteers to help validate the results of my HF channel simulator program. This is the version written in C. I am most interested in validating the statistics of the various fading paths - this needs to be obtained by several lengthy simulation runs. You would need to include your own statistic collection code, however, I will be happy to assist with this.

As a minimum to participate, you would need a reasonably fast computer (something in the 486/33 class), a C compiler, and spread sheet/statistics analysis software. You would need to be able to read C code and be able to make a few code changes. I don't think this is rocket science - most of the background have been discussed on this SIG.

If you are seriously interested, please e-mail - it is a short program so it can be e-mailed directly.

Any help in this regard would be greatly appreciated.

Thanks in advance and 73's

Johan, KC7WW

4. HF modulation schemes

From: "Johan Forrer"
Date: Wed, 14 Dec 1994 14:18:11 PST8PDT
To: HFSIG@tapr.org
Message-ID: <AC632344275@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

Just thought that I would pass on word of a gem of a book that I have been studying over the past few days:

"Principles and Practice of Multi-frequency telegraphy" by J.D Ralphs. Peter Peregrinus (1985).

I saw it on a post to "rec.radio.amateur.digital.misc" and checked it out from our library. If it was not so expensive (\$107), I would have considered buying a copy for my own reference.

This book is about "Piccolo", i.e. MFSK and covers some great topics. It has, for instance the clearest non-math approach to explaining matched filters that I have seen. It also goes into great lengths of describing HF channel simulation and is a very useful extension of the materials on HF channel simulation previously posted on this SIG. Also includes a chapter on how it rates against other modulation schemes, i.e. the 16-parallel DPSK MIL-188 modem (actually does somewhat better).

Food for thought.

73's

Johan, KC7WW

5. December QEX on Clover tests

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 15 Dec 1994 22:58:01 -0800

To: hfsig@tapr.org
Message-ID: <199412160658.WAA00937@servo.qualcomm.com>
Rendered body type: text/plain

The current issue of QEX has a rather interesting article on the results of some tests on Clover II using an HF channel simulator.

The character error rate vs Eb/N0 curves (no ARQ) are very telling. They compare several Clover data rates against a NATO HF modem standard, STANAG 4285. What's particularly interesting is that in both fading situations (1 and 2-path Rayleigh) STANAG 4285 outperformed CloverII by 10 dB or more for the same data rate (300 bps).

The author attributes much of the difference to STANAG 4285's use of a 3 KHz channel vs Clover's use of a 500 Hz channel. I think this illustrates very nicely the perils of narrowband thinking. Consider that STANAG 4285 not only requires less than 1/10 of the total transmitter power of Clover II for the same data rate, but it spreads it out over 6 times the bandwidth. This reduces the signal spectral density by more than 60x or 16 dB. That's a significant reduction in interference potential to others reusing the channel elsewhere.

The article also mentions MIL-STD-110A, saying that it's very similar to (but incompatible with) STANAG 4285. Since we have a CD-ROM database of MIL-STDs at work, I was able to print off a copy (81 pages). If someone is really interested in a copy, I could be persuaded to make some more and mail them out.

Phil

6. Re: [HFSIG:120] December QEX on Clover tests

From: frode@dxcern.cern.ch (Frode Weierud)
Date: Fri, 16 Dec 1994 09:42:17 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <9412160842.AA19133@dxcern.cern.ch>
Rendered body type: text/plain

Hi Phil,

The current issue of QEX has not yet arrived over here so I have not seen this article, but I am looking forward to study it.

Concerning STANAG 4285 and other NATO standards I have tried several times to get access to them, but so far without any success. Most of them are classified, some even quite heavily, and I have not even got a reply to some of my requests.

When I get the QEX article I will again try to approach the NATO standardisation people and see if I can get them to cooperate.

I would like very much to have a copy of MIL-STD-188/110A, "INTEROPERABILITY & PERFORMANCE STANDARDS FOR DATA MODEMS", so if you can mail me a copy I will be extremely grateful.

I will cover you costs for postage etc. Please let me know how I can reimburse you, with IRCs or a green bill in a letter. I would like to avoid sending you a cheque as I have to pay 10.- SFr for each cheque and that is hardly worth it for small amounts.

As Christmas is looming on the horizon I will take this opportunity to wish you a Merry Christmas and a Happy New Year.

73 Frode, LA2RL

```
*****
*      Frode Weierud           Phone   :      41 22 7674794      *
*      CERN, SL                Fax      :      41 22 7679185      *
*      CH-1211 Geneva 23       E-mail   :      frode@dxcern.cern.ch  *
*      Switzerland              or      weierud@cernvm.cern.ch  *
*****
```

7. Re: [HFSIG:120] December QEX on Clover tests

From: muphaus@cris.com (Marv Uphaus)
Date: Fri, 16 Dec 1994 03:01:47 -0600

10. Re: [HFSIG:122] Re: December QEX on Clover tests

From: Greg Jones <gjones@tenet.edu>
 Date: Fri, 16 Dec 1994 06:56:07 -0600 (CST)
 To: hfsig@tapr.org
 Message-ID: <199412161256.GAA02166@Gayle-Gaston.tenet.edu>
 Rendered body type: text/plain

Please feel free to use the tapr area on tcet.unt.edu
 Hopefully we will be moving it over to tapr.org., but a few more things are
 required before we do that.

Cheers - Greg

According to Marv Uphaus:

```
>
> On Fri, 16 Dec 94 01:00 CST, Phil Karn <karn@qualcomm.com> wrote:
>
> > Since we have a CD-ROM
> > database of MIL-STDs at work, I was able to print off a copy (81
> > pages). If someone is really interested in a copy, I could be
> > persuaded to make some more and mail them out.
> > ^^^^
> > Phil...
>
> Shame on you...!!! This is electronic media... Find an FTP site to post
> it on and just make an electronic copy of the 81 pages and let whoever
> wants a copy FTP it and print it out themselves... Since it's already
> on CD-ROM that ought to take you about 1 minute to perform...
>
> BTW, Johan, the HFSIG needs an ftp site to post information, results, etc.
> that everyone can get to... Any suggestions from anyone...
>
> Marv...
>
> -- Marv Uphaus -- muphaus@cris.com -- Ph: 205-343-9256 --
> -- U.S.Mail: 4031 Airport Blvd. #49 -- Mobile, AL 36608 --
> -- Packet Radio Address -- K4BVG @W4IAX.#MOBAL.AL.USA.NA --
>
```

11. An introduction and an apology

From: frode@dxccern.cern.ch (Frode Weierud)
 Date: Fri, 16 Dec 1994 14:32:47 +0100 (MET)
 To: hfsig@tapr.org
 Message-ID: <9412161332.AA24122@dxccern.cern.ch>
 Rendered body type: text/plain

Hi all,

As you have all seen that I exist, through my negligence
 of not checking the address field before hitting the Return
 key, I thought I at least could try to partly repair the
 damage by offering you a short introduction of myself.

I have been following the discussions on the list for about
 a month now and I have already learned a few things I did not
 know. I have been interested in HF transmission modes for
 more than 25 years and I have during those year been trying
 to decode many, if not all, of the exotic modes you find on
 the HF bands.

I am a licenced ham, LA2RL, but I have been living in the
 Geneva, Switzerland area for the past 22 years and for
 a large part of the time I have been unable to practice my hobby
 due to lack of a reciprocal licence. This made me into a
 "utility listener" which I still am, even if I rarely find
 the time to devote to this activity. Today I have a French
 temporary licence, F/LA2RL, but unfortunately I rarely have
 time for more than one QSO per week. I still hope to be
 able to find time to reactivate myself on the digital modes,
 and to again have some contacts on RTTY and Amtor and to try
 out the new Pactor mode.

I have had the intention of informing you all about a few
 papers that I have found of interest. Some of the references
 I had in mind others have already found, so I will not repeat
 those, but here are a few additional references:

You should try to have a closer look at a special issue of IEEE Journal on Selected Areas in Communications, Vol. SAC-5, No. 2, February 1987. This issue is dedicated to fading and multipath channel communications.

Here are two of the papers that I have found interesting:

- Seymour Stein; "Fading Channel Issues in System Engineering", pp.68-89.
- Kenneth Brayer; "HF Data Transmission: Lessons from the Past, Directions for the Future", pp.90-101.

I have also seen reference to two reports on single tone modems that it might be interesting to look at for those that would have access to them. They are:

- D.D. McRae, R.S. LeFever, and J.N. York; "HF modem feasibility demonstration", RADC-TR-2-48, Final Tech. Rep., Mar. 1981 - Oct. 1981.
- P. Monsen and L.C. Perry; "HF modem test and evaluation", RADC-TR-3-19, Final Tech. Rep., Jan. 1983.

Over Christmas I plan to look through my old files and if I find other papers of interest I will let you all know.

As I have already wished Phil a Merry Christmas and a Happy New Year I now want to make amends by wish you all happy festivities and a very fruitful and digital 1995.

73 Frode, LA2RL

```
*****
*      Frode Weierud      Phone   :      41 22 7674794      *
*      CERN, SL          Fax     :      41 22 7679185      *
*      CH-1211 Geneva 23  E-mail  :      frode@dxcern.cern.ch  *
*      Switzerland              or   weierud@cernvm.cern.ch  *
*****
```

12. Re: [HFSIG:123] Re: December QEX on Clover tests

From: muphaus@cris.com (Marv Uphaus)
Date: Fri, 16 Dec 1994 17:03:07 -0600
To: hfsig@tapr.org
Message-ID: <hoXykCysSgWD075yn@cris.com>
Rendered body type: text/plain

On Fri, 16 Dec 94 04:39 CST, Phil Karn <karn@qualcomm.com> wrote:

>The only problem is that the document is stored as an bitmap image on
>the CD-ROM and the only useful thing to do with it is to print it on a
>laser printer.

Phil...

Post the file as a "filename.ps" (post script file) to tcet.unt.edu...
Then anyone with access to a postscript printer (most people have a friend
with one) can print it out... Another option would be to put the file on a
WWW page in graphic form and then it can be looked at on-line... (but
pretty slow even at 14400...) If you post it as a file somewhere, I will
try some OCR software on it and see if we can convert it to ASCII and the
diagrams to .BMP files...

There are a lot of options besides printing it out in hardcopy... We
should NEVER have to do that again... That was what the paper companies
were afraid of years ago at the beginning of the computer revolution... So
far it hasn't happened but it's close at hand...!!!

73... Marv...

-- Marv Uphaus -- muphaus@cris.com -- Ph: 205-343-9256 --
-- U.S.Mail: 4031 Airport Blvd. #49 -- Mobile, AL 36608 --
-- Packet Radio Address -- K4BVG @W4IAX.#MOBAL.AL.USA.NA --

13. Re: [HFSIG:127] Re: December QEX on Clover tests

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 18 Dec 1994 00:10:35 -0800

To: muphaus@cris.com (Marv Uphaus)
Cc: hfsig@tapr.org
Message-ID: <199412180810.AAA00477@unix.ka9q.ampr.org>
Rendered body type: text/plain

The problem is that this is a proprietary packaged system, and there doesn't seem to be any way to get the data out of it in electronic form. I don't even know that it sends Postscript to the printer; it's an HP LaserJet III, so it could well be talking PCL for all I know.

By the way, to whoever maintains this mailing list: items are coming out with a "Reply-To:" header that points to the list. That makes it difficult to reply privately to the sender of a message. This is not good in my opinion.

Phil

14. Re: Achieving regulation by bandwidth

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Sun, 18 Dec 94 08:17:44 CST
To: hfsig@tapr.org, tcp-group@ucsd.edu, kd5i@kazak.nmsu.edu
Message-ID: <2355@k5yfw.ampr.org>
Rendered body type: text/plain

Seasons Greetings to you Karl and All.

In message <Pine.SUN.3.91.941216082204.24883C-100000@kazak> Karl writes:

>
> It's a pet theory of mine. I believe that a standard "rice box"
> HF rig such as my Kenwood TS-140 has decient filtering in the IF for a
> voice. It seems to be matched very well to the 1200 baud FM modulated
> waveform used for years on VHF. I have experimented with other pbbs
> sysop's on 10 meters using 1200 baud.

That would seem constant with my use of 1200 baud on 10 meter FM several years ago. But, I must comment that signals on the band very were stable. I suspect that I was getting little/few multi-path signals to cause distortion. However, at the first hint of signal degration, I would lose connectivity while the signal strength was still strong. At that time, I didn't know Multi-path signal distortion.

>
> The results were spectacular! We had such good radio paths that
> we could use maxframes of 2 and 3! The data rate was comperable to that
> acheived on vhf if the pbbs were in the same town talking direct to each
> other. I didn't get a ticket from the FCC for using data on a voice only
> frequency. I didn't get anything. When propagation got bad we quit.

As referenced above, it is (now to me) a well know fact that there is rapid fading observed on HF circuits that is the result of interference between a number of different waves that have been refracted from different portions of the ionosphere. Each refracted wave will arrive at the receive antenna as a slightly different time. This frequency or phase shift will mask a received data signal that has a baud rate generally above 150 baud.

With little or no multi-path signal distortion, 300 baud, and above is possible. However, in most cases you must be below 150-200 baud to prevent this problem from causing detection problems. In severe cases, the problem may exist down to 100 baud.

Since the above applies to AFSK and FSK, my assumption is that FM AFSK would also be affected.

I was running 1200 baud below 28.3 KHz.

>
> Reality says that 20 meters is where you need to be if you want
> to pass traffic on HF during a sunspot minimum. It has been shown that
> 300 baud packet radio is not as viable as Pactor and some other software
> designed for the limited bandwidth we allow ourselves in the "CW" portion
> of the band. I say if we go to 1200 baud in the "Phone" portion of the
> band we will stir up some discussion and show that packet radio works
> well on HF.

Please reference my comments above on multi-path signal distortion.

>
> I am willing to set up a 1200 baud packet forwarding frequency in
> the Extra Class portion of the band, say 14.54 MHz, lower sideband. I
> need at least another sysop willing to try this. The reason for doing
> this is 2 fold. It will measure how well 1200 baud works on 20 meters and
> provide some real data to show FCC at a future time when we want to argue
> for regulation by bandwidth used, not type of modulation.

I believe that someone else has reference the fact that data transmissions are only authorized, in areas under control of the F.C.C. between 14.00 MHz and 14.15 MHz on the 20 meter band.

Additionally, F.C.C. rules, Part 97.305 Authorized emission types, (97.307 Emission standards) authorizes only a "symbolic rate must not exceed 300 bauds, or for frequency shift keying, the frequency shift between the mark and space must not exceed 1 KHz."

Operating 1200 baud in the U.S. Phone band on 20 meters, if under the jurisdiction of the F.C.C. Rules Part 97 is prohibited. However, the F.C.C. might allow experimenting if properly solicited.

>
>
> 73, de karl k5di@k5di.nm.usa na k5di@acca.nmsu.edu
> DOS and Windows some say are bad software, but facts
> do not support this. Bill Gates has more than a
> Billion dollars!

There is a group who are interested in developing robust high speed data transmission protocols for HF. The thread(s) can be found by subscribing to "hfsig@tapr.org". For precise subscription directions, try sending E-Mail to listserv@tapr.org and in the text send only the work help...if this doesn't work, send the word info. But I'm willing to bet that someone like Greg Jones, will tell you just how to subscribe (are you reading this Greg?).

My burning desire is to be able to run 2400 BPS on HF like the \$7,500 to \$15,000 modems that the military use only using an off-the-shelf computer sound card. The military runs 2400 BPS on HF with better thru-put than ten AMTOR stations at the same S:N ratio. I also want to be able to interface TCP/IP (SMTP) from V/UHF AmprNet LANS to an HF AmprNet WAN.

73,

Walt@home.4.the.week-end

15. Re: [HFSIG:124] Re: December QEX on Clover tests

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sun, 18 Dec 1994 21:52:55 -0800
To: rs@ife.ee.ethz.ch
Cc: hfsig@tapr.org
Message-ID: <199412190552.VAA01428@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Is that CD-ROM with Mil-Stds publically available ? If so, do you know its source ?
>(some Mil-Stds are available on the net, however, no the more recent/relevant
>HF-modem standards).

My company subscribes to a commercial CD-ROM service for this stuff. I don't know much about it except the name: "Information Handling Services". Probably rather expensive, given the number of CD-ROMs.

Phil

16. FWD>RE>M4X- TMS320C4x Commu

From: "Rick Whiting" <Rick_Whiting@ATK.COM>
Date: 19 Dec 1994 11:10:39 -0600
To: "Post hfsig TAPR" <hfsig@tapr.org>
Cc: Darrell_Sawyer@gateway1.ATK.COM
Message-ID: <199412191853.MAA06689@ATK.COM>
Rendered body type: text/plain

Mail*Link(r) SMTP FWD>RE>M4X: TMS320C4x Communica

Date: 12/19/94 10:55 AM

From: Rick Whiting
In article <3cnkcp\$pcu@tilde.csc.ti.com>
Keith Larson <xkel@msg.ti.com> writes:

```
> M4X EVALUATION SOFTWARE NOW AVAILABLE
>
> A Multiprocessing TMS320C4x Communications and Debug Kernel
>
> What is M4X and how do I get a copy
> -----
> M4X is a complete reference design for a multiprocessing system of
TMS320C4x
> devices. All materials for M4X are contained in one self extracting file
> (M4X.EXE) which is available as freeware from the TMS320 BBS (713-274-2323)
> where the original source will be maintained. Under a freeware agreement
you
> are allowed to use, modify and freely distribute (IE no charge) this code
so
> long as you provide reference to the original authors.
>
> NOTE: At this time the M4X reference design is intended for evaluation
use
> only and will NOT be posted to TI's BBS ftp site until after the evaluation
> period is completed.
>
> Who might be interested in M4X
> -----
> This application is best suited for builders and hackers. To use this
> application you will need to build your own interface card (less than $5 in
> parts) and target system (more than $5) from the descriptions given in the
M4X
> documentation. M4X is intended for those individuals who are willing to
> participate in the development of an open, user supported standard. If you
> make significant advancements, please contact the original M4X developers
to
> have your ideas incorporated.
>
> Support
> -----
> M4X will NOT be supported by the TMS320 Hotline. Instead, users should
send
> messages, comments and code to other M4X users through the TMS3230 BBS, or
an
> internet group, when and if it should form.
>
> What is contained in M4X.EXE
> -----
> M4X.EXE is a self extracting executable which contains several
executables,
> source files and documentation. The main files, and what they do, are
given below.
>
> M4X.ASM - Assembly source code for TMS3204x runtime kernel. Provides
> bi-directional split mode DMA and commport control for packetized
> routing of commands and data.
>
> M4XD.EXE - Multiprocessing M4X debugger interface which runs in a
> standard DOS window. Multiple DOS windows and debug nodes are
> allowed. Sorry no source code can be made available for this
> debugger!
>
> MANDEL40.EXE - Very fast multiprocessor version of the Mandelbrot set.
> Runs a full screen SVGA capable of 640x480 256 colors
>
> MEMVIEW.EXE - Multiprocessor memory viewer which runs independent of other
M4X
> tasks. You can use this to asynchronously read and display data
>
> blocks in a running system.
>
> Documentation - ICSPAT Conference paper (in postscript) and M4X
documentation
> are included. The M4X kernel and applications were first
introduced
> and demonstrated at the DSP World, ICSPAT conference in Dallas in
>
> October 1994.
>
> Theory of Operation
```

```

> -----
> The M4x system utilizes the commport bootload feature of the TMS320C4x to
> initialize an unknown system of processors in a wavefront bootload process.
>
> The bootload process detects all valid connections, no connects and null
> outputs and builds an interconnection matrix from the bootup data.
>
> When loaded the M4X kernel provides several low level functions which
> are used to manage a multiprocessor system. The functions include getmem,
> putmem, matrix management, call, branch, run, halt and singlestep.
>
> The host communicates with the system through I/O driver and target
> command processing code. The I/O driver code interfaces with the hardware
to
> provide token, data strobe and fifo control. The target command software
is
> then responsible for sending and receiving packetized command streams from
> the system.
>
> The application code provided shows a user how to interface applications
to a
> system in a general purpose and easy to use manner. Other important
system
> level functions which are provided include the system bootloader, coff file
> loader, data transfer protocols, and simple run, stop and debug.
>
> Two I/O circuits are given which can be used to connect a TMS320C4x
commport
> to a PC's printer port (needs to be bi-directional). These simple circuits
> are designed to provide the necessary buffering and logic through software
> control to achieve a bi-directional general purpose link to a TMS320C4x
commport.
> Since all of the printer port signals are uni-directional, simple
buffering
> techniques can be used to provide links of unlimited distance. This
interface
> is also easily converted to other processors such as the TMS320C3x.
>
> Conclusion
> -----
> M4X has been developed to provide a starting point for the low level BIOS
> framework of a multiprocessing system. When coupled with higher levels of
> functionality there are many possibilities. At this time it is definitely
NOT
> complete and still needs lots of work. It just takes time and effort!
>
> Best regards,
> Keith Larson
>
> Disclaimers
> -----
> They said it could'nt be done, so I did it!
>
> If you follow the rules that everybody else follows, you are
> doomed to repeat the actions of everybody else!
>
> There is no such thing as a free lunch!
>
> I have been known to make a mistake or two, this is how I learn!
>
> If you see a bug.. Kill it first! Then give me a call.
>
> Yes, I also created the C26 DSK... and I am very busy.
>

```

```

-----
| Rick Whiting                Office Phone 612-931-5344      |
| 5780 Rosewood Lane N.      Home Phone 612-550-1213      |
| Plymouth, MN 55442         Fax 612-931-5735             |
| U.S.A.                     E-mail rick_whiting@atk.com    |
|                             Packet W0TN @ WB0GDB.MN.USA.NOAM |
-----

```

17. FWD>RE>New 56002 Evaluation

From: "Rick Whiting" <Rick_Whiting@ATK.COM>
Date: 19 Dec 1994 11:08:41 -0600

To: "Post hfsig TAPR" <hfsig@tapr.org>
Cc: Darrell_Sawyer@gateway1.ATK.COM
Message-ID: <199412191943.NAA07633@ATK.COM>
Rendered body type: text/plain

Mail*Link(r) SMTP FWD>RE>New 56002 Evaluation Mod

Date: 12/19/94 10:55 AM
From: Rick Whiting
In article <kirk.787276719@mustang18>
kirk@rtsg.mot.com (Michael J. Kirk) writes:

> Please excuse me if this info has
> already been posted. I just wanted to share
> this with anyone who might be interested in a low
> cost DSP evaluation tool. I found it listed in the
> October 1994 issue of DSP & Multimedia Technology.
>
> WHO: Motorola Semiconductor Products Sector
> Digital Signal Processor Division
> (512)-891-3717
>
> WHAT: A DSP56002 evaluation module containing:
> Hardware:
> Fully assembled and tested printed circuit board with
> DSP56002, 24 bit fixed point DSP, 40MHz
> 32k X 24 external SRAM
> Crystal Semiconductor MWAVE CS4215 Stereo Codec
> MC7805ACT voltage regulator
> MCHC705K1 microcontroller for RS232 to OnCE interface
> MC33078 preamp for analog buffering
> Jacks for stereo inputs, outputs and headphones
> Documentation for DSP56002, CS4215, ASSEMBLER, and
> DEBUGGER plus board schematics
> Software:
> Motorola's DSP5600x cross assembler for PCs
> Domain Technologies debug software with (DOS?)
> based windowed interface, SYMBOLIC DEBUGGING!
> Installation instructions
> Demo software and self test files
> Requirements:
> User must provide:
> Power supply (7-9V AC or DC) with spade terminals or
> 2.1 mm plug
> RS-232 cable with DB-9 connectors
> IBM PC compatible (-386 class or higher) running
> MS-DOS with 19200 baud serial port
> Options:
> 32k X 8 EEPROM for bootstrapping and stand-alone operation
> second RS-232 connector for access to DSP56002 SCI port
> second crystal for other sample rates, e.g. 44.1kHz
>
> HOW MUCH: US\$ 149.95 + shipping and applicable taxes
> Includes everything in WHAT above.
>
> WHERE: This offer may not be available outside the US
> check with your local Motorola sales office.
>
> WHEN: Available now.
>
> WHY: To get more people using Motorola DSP's
> To offer an alternative to TI's US\$99 evaluation kit
>
> All for now...
>
> Mike Kirk
> Senior Engineer
> Motorola, Inc.
> Cellular Infrastructure Group
> 1475 W. Shure Drive
> Arlington Heights, IL 60004

```
-----
| Rick Whiting           Office Phone 612-931-5344
| 5780 Rosewood Lane N. Home Phone 612-550-1213
| Plymouth, MN 55442    Fax          612-931-5735
| U.S.A.                E-mail       rick_whiting@atk.com
|                        Packet       W0TN @ WB0GDB.MN.USA.NOAM
-----
```

18. Re: [HFSIG:129] Re: Achieving regulation by bandwidth

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 19 Dec 1994 14:03:50 -0800
To: k5yfw@k5yfw.ampr.org
Cc: hfsig@tapr.org
Message-ID: <199412192203.OAA02330@unix.ka9q.ampr.org>
Rendered body type: text/plain

> With little or no multi-path signal distortion, 300 baud, and
> above is possible. However, in most cases you must be below
> 150-200 baud to prevent this problem from causing detection
> problems. In severe cases, the problem may exist down to 100
> baud.

So don't use a high baud rate. This implies multiple bits per transmitted symbol (channel state change). This can be done either orthogonally or non-orthogonally. Orthogonal simply means that all of the possible channel states can be detected independently, without mutual interference. Binary FSK with a sufficiently wide shift for the baud rate is the classic example; you can build a "tone filter" that responds just to mark, and another one that responds only to space.

But nothing limits FSK to just two tones. If you used, say 256 distinct tones then you could encode the 256 possible combinations of an 8-bit data symbol as one of the 256 tones. This would be 256-ary FSK. The general scheme is called m-ary FSK and has long been known to be one of the best schemes around for fading HF channels. The higher the value of m, the better the performance -- assuming you can spare the bandwidth.

The most common NON-orthogonal signalling scheme for multiple bits per transmitted symbol is QAM. This maps a n-bit data symbol into 2^n possible combinations of carrier amplitude and phase. QAM's big advantage is very good bandwidth efficiency for large values of n, which is why it's universal in telephone line modems. But big values of n mean very closely packed signal constellations, and relatively little noise is required to move one received signal state to another one nearby and cause an error. So the bandwidth efficiency of QAM comes at the price of a much higher required S/N ratio. This is again okay for telephone modems where the S/N ratio is usually 30 dB or more, but (in my opinion) it is completely inappropriate for noisy, interference-prone radio channels.

So once again we're back to the conclusion that to build a decent HF modem that provides a good link with minimum transmitter power, we really need to do something to relax the bandwidth limits.

Phil

19. Re: [HFSIG:133] Re: Achieving regulation by bandwidth

From: "Johan Forrer
Date: Mon, 19 Dec 1994 15:07:05 PST8PDT
To: hfsig@tapr.org
Message-ID: <B3F06525272@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Phil Karn wrote:

>So don't use a high baud rate. This implies multiple bits per
>transmitted symbol (channel state change). This can be done either
>orthogonally or non-orthogonally. Orthogonal simply means that all of
>the possible channel states can be detected independently, without
>mutual interference. Binary FSK with a sufficiently wide shift for the
>baud rate is the classic example; you can build a "tone filter" that
>responds just to mark, and another one that responds only to space.

>But nothing limits FSK to just two tones. If you used, say 256
>distinct tones then you could encode the 256 possible combinations of
>an 8-bit data symbol as one of the 256 tones. This would be 256-ary
>FSK. The general scheme is called m-ary FSK and has long been known
>to be one of the best schemes around for fading HF channels. The
>higher the value of m, the better the performance -- assuming you can
>spare the bandwidth.

As supporting arguments to above, please see J.D. Ralphs's book: Principles and practice of multi-frequency telegraphy. Peter Peregrinus, ISBN 0-86341-022-7. This book includes extensive chapters on how such a system is engineered around the symbol set and signaling rate to

combat the properties of the HF channel. It also compares this m-FSK to the MIL-STD-188A 16-tone parallel modems and comes out a nose ahead of the pack. The book also describes a simple synchronous detector that would be quite simple to implement on our present DSP platforms. If you are interested in using an HF channel simulator this reference also contains very useful stuff.

>The most common NON-orthogonal signalling scheme for multiple bits per
>transmitted symbol is QAM. This maps a n-bit data symbol into 2^n
>possible combinations of carrier amplitude and phase. QAM's big
>advantage is very good bandwidth efficiency for large values of n,
>which is why it's universal in telephone line modems. But big values
>of n mean very closely packed signal constellations, and relatively
>little noise is required to move one received signal state to another
>one nearby and cause an error. So the bandwidth efficiency of QAM
>comes at the price of a much higher required S/N ratio. This is again
>okay for telephone modems where the S/N ratio is usually 30 dB or
>more, but (in my opinion) it is completely inappropriate for noisy,
>interference-prone radio channels.

The possibilities of QAM is quite attractive. Its signaling constellation is shown to have distinct robustness over other alternative schemes of phase/amplitude signaling especially under conditions that include conditions of phase jitter (please see the Foschini paper that I posted a while ago).

I have been thinking about the possibilities of QAM especially after reading Ralphs' book. He describes a simple procedure that is based on a simple synchronization sequence, and once lock is achieved, the PLL stays locked during the remainder of the frame (he gives supportive arguments of its merits). If the signalling rate is chosen appropriately, the system will tolerate a considerable amount of phase distortion. I would like to invite further comments on this type of idea. I am not sure why, if this is such a scheme is this good, why have we not seen more general acceptance?

>So once again we're back to the conclusion that to build a decent HF
>modem that provides a good link with minimum transmitter power, we
>really need to do something to relax the bandwidth limits.
>Phil

Ditto - it is inevitable, however, even 1kHz would take us a long way and this is within part 97.

73's

Johan, KC7WW

20. Re: [HFSIG:133] Re: Achieving regulation by bandwidth

From: Greg Jones <gjones@tenet.edu>
Date: Mon, 19 Dec 1994 17:58:16 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199412192358.RAA13787@Gayle-Gaston.tenet.edu>
Rendered body type: text/plain

According to Phil Karn:

> So once again we're back to the conclusion that to build a decent HF
> modem that provides a good link with minimum transmitter power, we
> really need to do something to relax the bandwidth limits.

How wide are we talking Phil ?

Which bandwidth limits are you discussing?

Greg

21. Re: [HFSIG:134] Re: Achieving regulation by bandwidth

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 19 Dec 1994 18:06:42 -0800
To: FORRERJ@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199412200206.SAA03125@unix.ka9q.ampr.org>
Rendered body type: text/plain

>The possibilities of QAM is quite attractive. Its signaling
>constellation is shown to have distinct robustness over other
>alternative schemes of phase/amplitude signaling especially under
>conditions that include conditions of phase jitter (please see the Foschini
>paper that I posted a while ago).

I'm not convinced. QAM requires phase-coherent detection, but the HF channel is inherently phase random. A Rayleigh fading channel model has a Rayleigh probability distribution on amplitude (hence the name) and a uniform distribution of phase between 0 and 2π - i.e., the received phase is completely random and can carry no information due to the constantly changing and unpredictable propagation path lengths. Although you might be able to get away with phase modulation on some particularly benign HF channel, especially if there are only a small number of path components, it sure doesn't seem like a wise thing to rely on in your design.

So this implies noncoherent demodulation, i.e., just looking at signal energy and ignoring phase. m-ary FSK can be demodulated this way. As you increase m towards infinity you not only gain back everything you lost by going noncoherent, but you also approach the -1.6dB Shannon limit!

This stuff actually works. Although the details and implementation are quite different, Qualcomm's CDMA system uses 64-ary modulation on the reverse (mobile-to-cell) link in order to make up for the lack of a coherent carrier phase reference over the Rayleigh-faded mobile radio channel. The orthogonal codes are Walsh functions over time instead of collections of sine waves over frequency, but the same principles apply.

--Phil

22. Re: [HFSIG:135] Re: Achieving regulation by bandwidth

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Mon, 19 Dec 1994 18:27:53 -0800
To: gjones@tenet.edu
Cc: hfsig@tapr.org
Message-ID: <199412200227.SAA03205@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Which bandwidth limits are you discussing?

Those on the bottom half of each HF band.

>How wide are we talking Phil ?

As wide as possible, as long as we stay in the band. For short-term practical reasons, a good start would be 2-3 Khz. That fits in a SSB filter so existing HF transceivers could be used without modification by simply writing the appropriate DSP code to drive them with baseband audio.

>Which bandwidth limits are you discussing?

Those on the bottom half of each HF band, where "data" is allowed.

Phil

23. TAPR 1995 Annual meeting

From: "Johan Forrer
Date: Tue, 20 Dec 1994 08:57:26 PST8PDT
To: HFSIG@tapr.org
Message-ID: <B50DDD479AB@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

I hope that everyone is giving this meeting serious thought? If you are interested in DSP, and you should by now (?). This will be a good place to be. This will be an excellent opportunity to get first-hand impressions on what are being done and also an opportunity to attend the seminars. I am looking forward to hearing Phil's presentation on "Error correction coding". This is, in my opinion, a key component in a modern communication system. It's just a shame that the two planned seminars are in a time conflict as I would have loved attending the DSP-93 seminar too. Is there any possibility in re-arranging these venues?

I have also been approached to present a paper, and though, public speaking is not my favorite passtime, I will consider a topic. I have sufficient materials on two topics:

1) An implementation of PACTOR on a PC. This could look at the Pactor protocol, its good and not-so-good aspects, how frame sync is achieved and maintained through brute-force algorithms, memory ARQ, and how this is integrated into the sound card's DSP.

2) The "HFSIG" HF channel simulator. This could include the Watterson model, a summary of various bits and pieces as contributed on this SIG, and my simple PC/DSP-sound card implementation.

Suggestions?

73's, Johan KC7WW

24. Re: [HFSIG:133] Re: Achieving regulation by bandwidth

From: "Johan Forrer
Date: Tue, 20 Dec 1994 09:23:38 PST8PDT
To: hfsig@tapr.org
Message-ID: <B514DCB4577@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Phil Karn wrote:

>>The possibilities of QAM is quite attractive. Its signaling
>>constallation is shown to have distinct robustness over other
>>alternative schemes of phase/amplitude signaling especially under
>>conditions that include conditions of phase jitter (please see the
>>Foschini >paper that I posted a while ago).

>I'm not convinced. QAM requires phase-coherent detection, but the HF
>channel is inherently phase random. A Rayleigh fading channel model
>has a Rayleigh probability distribution on amplitude (hence the name)
>and a uniform distribution of phase between 0 and 2pi - i.e., the
>received phase is completely random and can carry no information due
>to the constantly changing and unpredictable propagation path lengths.
>Although you might be able to get away with phase modulation on some
>particularly benign HF channel, especially if there are only a small
>number of path components, it sure doesn't seem like a wise thing to
>rely on in your design.

>So this implies noncoherent demodulation, i.e., just looking at signal
>energy and ignoring phase. m-ary FSK can be demodulated this way. As
>you increase m towards infinity you not only gain back everything you
>lost by going noncoherent, but you also approach the -1.6dB Shannon
>limit!

Yes that is quite true if you consider doing things the way its done
on telco circuits,

i.e. "matched filter" == coherent demodulation + integrate&dump +
thresholding.

It would be really valuable getting hold of Ralph's book and see the
logic of relaxing the coherent demodulation requirement. He calls it
"quasi"

coherent detection and still uses matched filters, though at passband, i.e.
filters engineered and spaced as matched filters at passband instead of at
baseband. I am sure that using this scheme I can implement DPSK + at least
some form of amplitude modulation like being done on Clover. Although this
is not quite the same as true QAM by definition, we can still
achieve multiple bits per baud and also enjoy the bandwidth efficiency
benefits. Does this make any sense - is it worth looking into?

---further stuff deleted ---

I think we are making progress - keep up the good work.

73's, Johan, KC7WW

25. Re: [HFSIG:138] TAPR 1995 Annual meeting

From: Greg Jones <gjones@tenet.edu>
Date: Tue, 20 Dec 1994 11:55:27 -0600 (CST)

To: hfsig@tapr.org
Message-ID: <199412201755.LAA08905@Gayle-Gaston.tenet.edu>
Rendered body type: text/plain

According to Johan Forrer

FL:

> I hope that everyone is giving this meeting serious thought?
> If you are interested in DSP, and you should by now (?). This
> will be a good place to be. This will be an excellent opportunity to
> get first-hand impressions on what are being done and also an
> opportunity to attend the seminars. I am looking forward to hearing
> Phil's presentation on "Error correction coding". This is, in my
> opinion, a key component in a modern communication system. It's just a
> shame that the two planned seminars are in a time conflict as I would have
> loved attending the DSP-93 seminar too. Is there any possibility
> in re-arranging these venues?

Hum -- interesting comment. I'll talk to Phil, Bob, and Mel and see what we might be able to do. I should have realized this when Mel and myself sat down and did the initial schedules.

The reason for finishing up around noon is to give folks a chance to fly out the rest of the afternoon. If a session is from 1pm till 4 or 5pm will people stay and be able to get their flight out. or maybe shorten Phil's a tad and Bob's a tad, but people will still want to go to lunch sometime :-)
You can see the problems.

Greg

26. Re: [HFSIG:133] Re: Achieving regulation by bandwidth

From: "Johan Forrer"
Date: Tue, 20 Dec 1994 10:00:58 PST8PDT
To: hfsig@tapr.org
Message-ID: <B51ED1660BE@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

Phil, it may help a great deal and be very beneficial to a lot of us if you could be so kind and, in a paragraph or so, explain what is meant by:

Channel capacity (entropy),

Spectral occupancy,

Bandwidth efficiency,

and how all this eventually relates to the Shannon limit.

I'm certain that understanding these definitions will help us understand better what we are trying to achieve. From your past postings, you have a great talent for making such stuff clear.

Thanks a lot in advance.

73's, Johan, KC7WW

27. Re: [HFSIG:138] TAPR 1995 Annual meeting

From: Hugh Shane <shane@mdd.comm.mot.com>
Date: Tue, 20 Dec 1994 10:28:35 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941220095521.28112U-100000@daffyduck>
Rendered body type: text/plain

On Tue, 20 Dec 1994, Johan Forrer FL wrote:

>
> I have also been approached to present a paper, and though, public speaking
> is not my favorite passtime, I will consider a topic. I have sufficient
> materials on two topics:
>
>
> Suggestions?
>

How about a summary of the issues that have been debated on this net? For instance:

* Is more bandwidth for digital HF channels necessary/desirable/feasible?
* What modulation techniques are presently under evaluation? What are the pros and cons of each.
* DSP platforms for HF signal processing: pros and cons of different makes, writing portable code.

All of this may be self-evident to those who have been following this group for a while, but to outsiders a summary could be instructive.

Hugh

28. Re: [HFSIG:142] Re: TAPR 1995 Annual meeting

From: "Johan Forrer
Date: Tue, 20 Dec 1994 12:20:31 PST8PDT
To: hfsig@tapr.org
Message-ID: <B5440926DB3@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

On Tue, 20 Dec 1994, Hugh Shane wrote:

>>
>> I have also been approached to present a paper, and though, public speaking
>> is not my favorite passtime, I will consider a topic. I have sufficient
>> materials on two topics:
>>

>>
>> Suggestions?
>>

>How about a summary of the issues that have been debated on this net? For
>instance:

>* Is more bandwidth for digital HF channels necessary/desirable/feasible?
>* What modulation techniques are presently under evaluation? What are the
>pros and cons of each.

I'm afraid that I have to pass this one to someone more capable as
I don't think that I can do justice that these deserve. How about you Hugh?

>* DSP platforms for HF signal processing: pros and cons of different
>makes, writing portable code.
>All of this may be self-evident to those who have been following this
>group for a while, but to outsiders a summary could be instructive.

I can consider this one, if it is not taken as a cheap shot at the
TAPR-93. How does others feel about it? Let us hear your suggestions.

73's

Johan, KC7WW

29. HF Modulation Modes

From: Alan Bloom <alanb@eagle.sr.hp.com>
Date: Tue, 20 Dec 1994 14:17:39 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <9412202217.AA06592@eagle.sr.hp.com>
Rendered body type: text/plain

I haven't posted to a list before, so let's see if this works...

In all the discussion about the best modulation format for HF, I'm surprised nobody has mentioned OFDM. Orthogonal frequency-division multiplex is proposed for Digital Audio Broadcasting and digital TV in Europe, although it is not so well known in the US.

The idea is to split the signal up into a large number (like 128 or 1024) equally-spaced carriers, each modulated at the same symbol rate. I believe the DAB standard proposes that each carrier be modulated with QPSK, but you can use any modulation in which each symbol can be represented by a phase and amplitude. That includes the many forms of QAM, PSK, PI/4 DQPSK, on-off keying, whatever.

For example, If you use 128 carriers, the symbol rate is reduced by

a factor of 128, greatly reducing the probability of inter-symbol interference due to multipath propagation. Adding forward error correction and time and frequency interleaving reduces the probability of bit errors due to selective fading on one or more carriers.

The reason the mode is called "orthogonal" FDM is that the carrier frequency spacing is 1/symbol period, which theoretically eliminates interference from all unwanted carriers. This is easiest to see in the time domain. For a symbol rate of 1 symbol/s, the carriers will be spaced 1 Hz apart. During each symbol period, the detector for each carrier receives a certain number of RF cycles of the desired carrier. During that same time period, each unwanted carrier has a number of cycles that differ from the desired carrier by an integer. Convolving two sine waves that differ by an integer number of cycles gives zero result.

You could implement OFDM by building 128 RF oscillators, 128 I/Q modulators, and the associated controlling hardware. But there's a much simpler way, and here's the clever part of OFDM:

I mentioned that the signal consists of a series of equally-spaced RF signals, each characterized by an amplitude and a phase. If you do an inverse DFT on that data, you come out with the time samples of the composite signal! At the kinds of data rates we are talking about, an inexpensive DSP would be plenty fast enough. For example, a 12.5 MHz ADSP21xx can do a 256-point DFT in 0.59 ms. So we're talking about a \$10 DSP and a single I/Q modulator running at very low symbol rates.

For example, 128 QPSK carriers spaced 7.5 Hz apart (7.5 baud) gives 1920 bits per second within the 1 kHz bandwidth limit. The DSP has $1/7.5 = 133$ ns to do all its calculations.

The choice of modulation type for the carriers affects only the amplitude/phase list for each symbol. So it would be easy to change modulation types to respond to different propagation conditions, ala Clover. By including the capability to do various size FFTs, different symbol rates could be implemented using the same bit rate within the same bandwidth. i.e. use very slow symbol rates (large FFT) under fast Rayleigh fading conditions and faster symbol rates (small FFT) with slow flat fading.

Alan Bloom N1AL

30. Re: [HFSIG:144] HF Modulation Modes

From: "Johan Forrer"
Date: Tue, 20 Dec 1994 15:08:58 PST8PDT
To: hfsig@tapr.org
Message-ID: <B570F42212D@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Alan Bloom wrote:

>I haven't posted to a list before, so let's see if this works...

Welcome Al - glad you found us.

--some stuff deleted ---

>The idea is to split the signal up into a large number (like 128 or
>1024) equally-spaced carriers, each modulated at the same symbol rate.
>I believe the DAB standard proposes that each carrier be modulated
>with QPSK, but you can use any modulation in which each symbol can be
>represented by a phase and amplitude. That includes the many forms of
>QAM, PSK, PI/4 DQPSK, on-off keying, whatever.

If I understand it correctly, I do believe that the 39-tone parallel MIL-STD-188 modem that have been mentioned on this group, is based on a similar principle - if I understand it correctly?

-- more stuff deleted --

>I mentioned that the signal consists of a series of equally-spaced
>RF signals, each characterized by an amplitude and a phase. If you
>do an inverse DFT on that data, you come out with the time samples
>of the composite signal! At the kinds of data rates we are talking
>about, an inexpensive DSP would be plenty fast enough. For example,
>a 12.5 MHz ADSP21xx can do a 256-point DFT in 0.59 ms. So we're
>talking about a \$10 DSP and a single I/Q modulator running at very

>low symbol rates.

>For example, 128 QPSK carriers spaced 7.5 Hz apart (7.5 baud) gives
>1920 bits per second within the 1 kHz bandwidth limit. The DSP
>has $1/7.5 = 133$ ns to do all its calculations.

>The choice of modulation type for the carriers affects only the
>amplitude/phase list for each symbol. So it would be easy to
>change modulation types to respond to different propagation
>conditions, ala Clover. By including the capability to do various
>size FFTs, different symbol rates could be implemented using the
>same bit rate within the same bandwidth. i.e. use very slow
>symbol rates (large FFT) under fast Rayleigh fading conditions
>and faster symbol rates (small FFT) with slow flat fading.

This is interesting - one question: did we overcome the need for
coherent detection, or does it matter?

73's

Johan, KC7WW

31. Re: [HFSIG:145] Re: HF Modulation Modes

From: "Johan Forrer"
Date: Tue, 20 Dec 1994 16:25:36 PST8PDT
To: hfsig@tapr.org
Message-ID: <B5855F80B9A@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Some further food for thought:

Here are some further details on the 16-tone MIL-STD-188C modem
and a comparative M-FSK system, just for interest sake.

This MIL-STD modem uses 16 simultaneous tones, each modulated at 75 baud
(13.3 ms elements) in two or four phases giving a maximal rate of
 $2 \times 75 \times 16 = 2400$ bit/s in 3 kHz. The tones are spaced at 110 Hz, orthogonal
for the matched filter integration period of 9.1 ms, which allows for a
guard time of ± 2 ms to cater for a limited degree of sych. error.
The provision of guard periods causes an effective power loss of
1.75 dB. Assuming each tone quadrature modulated, the normalized bandwidth
is 3 Hz/bit/s. However, extending this to 16 parallel tones, this yields
0.87 Hz/bit/s, which is pretty close to the ultimate.

I have some further specifications on its BER, but unfortunately not
with me at the moment. Walt, K5YFW, could perhaps attest how these
have been doing on the military HF nets. This is living proof that
phase modulation is worth considering on HF.

One consideration when dealing with such a multi-tone signaling system,
that the power rating of the transmitter is rated on the assumption that a
single unmodulated carrier is radiated, however, with multiple tones
the output power is reduced by the square of the number of tones. Also
remember that, when dealing with the combination of such multiple tones
that a "noise-like" waveform results with possibly high peak/RMS ratio
(i.e. crest factor) that may cause distortion or inter-modulation effects.

According to Ralph's book, a 6-channel 6-FSK (i.e. six 6-FSK systems
paralleled using similar bandwidth) outperforms this MIL-STD-188C
modem. Sounds interesting?

73's

Johan, KC7WW

32. please subscribe

From: bart@wb6hbk.ampr.org (Bart Rowlett)
Date: Tue, 20 Dec 1994 17:03:23 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0rKF5d-0004bNC@wb6hbk.ampr.org>
Rendered body type: text/plain

subscribe

33. Re: [HFSIG:136] Re: Achieving regulation by bandwidth

From: Stephen Hall <shall@rain.org>
Date: Tue, 20 Dec 1994 20:10:25 -0800 (PST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.941220200614.7789C-100000@coyote.rain.org>
Rendered body type: text/plain

Greetings Phil, Steve here, WM6P. Have not seen you since my visit to Qualcomm. I recently left Harris Corp but may be able to get you in touch with some of the guys that developed the Harris serial and parallel tone HF modems. They seem to know as much as anyone at this point that I have run into.

73, Steve

34. [HFSIG:145] Re: HF Modulation Modes

From: BLOOM_ALAN/HP5300_R0@opnmail3.corp.hp.com
Date: Tue, 20 Dec 94 20:20:00 -0800
To: hfsig@tapr.org
Message-ID: <"d0eqoS-0000000000*"@MHS>
Rendered body type: text/plain

Johan KC6WW said:

>This is interesting - one question: did we overcome the need for
>coherent detection, or does it matter?

On the receiving end, I presume you would just run a DFT on the incoming sampled data which would output the frequency and phase of each carrier. It is effectively coherent detection, since the phase is known.

Alan N1AL

35. Re: [HFSIG:144] HF Modulation Modes

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Tue, 20 Dec 94 23:18:59 CST
To: hfsig@tapr.org
Message-ID: <2378@k5yfw.ampr.org>
Rendered body type: text/plain

Gee Alan, kinda sounds like the 16 & 39 parallel tone modem formats in MIL-STD-188-110c. 16 or 39 parallel tones, with QPSK, inter-leaving and Reed-Soloman coding for FEC. In the 39 tone modem, a 40th tone has no modulation and is called the Doppler Tone. If phase shift is detected on it, the modem knows than inter-symbol interference due to multipath propagation is taking place. The baud rate on the 39 tone modem is less than 50 baud and the bandwidth of the signal is 3 KHz. The transmitted "data thrupt rate (their terminology) is 2400 BPS. The actual transmitted bit rate is 3200-3600...I've forgotten and don't want to dig into my files tonight. The CLOVER II is a 4 parallel tone modem (and a few other things) and runs 850-900 BPS.

Walt@home.2.nite

In message <9412202217.AA06592@eagle.sr.hp.com> you write:

> I haven't posted to a list before, so let's see if this works...

>

> In all the discussion about the best modulation format for HF, I'm
> surprised nobody has mentioned OFDM. Orthogonal frequency-division
> multiplex is proposed for Digital Audio Broadcasting and digital TV
> in Europe, although it is not so well known in the US.

>

> The idea is to split the signal up into a large number (like 128 or
> 1024) equally-spaced carriers, each modulated at the same symbol rate.
> I believe the DAB standard proposes that each carrier be modulated
> with QPSK, but you can use any modulation in which each symbol can be
> represented by a phase and amplitude. That includes the many forms of
> QAM, PSK, PI/4 DQPSK, on-off keying, whatever.

>

> For example, If you use 128 carriers, the symbol rate is reduced by
> a factor of 128, greatly reducing the probability of inter-symbol
> interference due to multipath propagation. Adding forward error
> correction and time and frequency interleaving reduces the probability
> of bit errors due to selective fading on one or more carriers.

>

> The reason the mode is called "orthogonal" FDM is that the carrier
> frequency spacing is 1/symbol period, which theoretically eliminates
> interference from all unwanted carriers. This is easiest to see in
> the time domain. For a symbol rate of 1 symbol/s, the carriers will

> be spaced 1 Hz apart. During each symbol period, the detector for
> each carrier receives a certain number of RF cycles of the desired
> carrier. During that same time period, each unwanted carrier has a
> number of cycles that differ from the desired carrier by an integer.
> Convolving two sine waves that differ by an integer number of cycles
> gives zero result.
>
> You could implement OFDM by building 128 RF oscillators, 128 I/Q
> modulators, and the associated controlling hardware. But there's a
> much simpler way, and here's the clever part of OFDM:
>
> I mentioned that the signal consists of a series of equally-spaced
> RF signals, each characterized by an amplitude and a phase. If you
> do an inverse DFT on that data, you come out with the time samples
> of the composite signal! At the kinds of data rates we are talking
> about, an inexpensive DSP would be plenty fast enough. For example,
> a 12.5 MHz ADSP21xx can do a 256-point DFT in 0.59 ms. So we're
> talking about a \$10 DSP and a single I/Q modulator running at very
> low symbol rates.
>
> For example, 128 QPSK carriers spaced 7.5 Hz apart (7.5 baud) gives
> 1920 bits per second within the 1 kHz bandwidth limit. The DSP
> has $1/7.5 = 133$ ns to do all its calculations.
>
> The choice of modulation type for the carriers affects only the
> amplitude/phase list for each symbol. So it would be easy to
> change modulation types to respond to different propagation
> conditions, ala Clover. By including the capability to do various
> size FFTs, different symbol rates could be implemented using the
> same bit rate within the same bandwidth. i.e. use very slow
> symbol rates (large FFT) under fast Rayleigh fading conditions
> and faster symbol rates (small FFT) with slow flat fading.
>
> Alan Bloom N1AL
>
>

36. Re: [HFSIG:146] Re: HF Modulation Modes

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Tue, 20 Dec 94 23:39:45 CST
To: hfsig@tapr.org
Message-ID: <2381@k5yfw.ampr.org>
Rendered body type: text/plain

In message <B5855F80B9A@frl.orst.edu> you write:

[stuff deleted]
> I have some further specifications on its BER, but unfortunately not
> with me at the moment. Walt, K5YFW, could perhaps attest how these
> have been doing on the military HF nets. This is living proof that
> phase modulation is worth considering on HF.
[stuff deleted]
> 73's
>
> Johan, KC7WW

Do you "guys" remember all those "Sadamn" cartoons during Desert
Shield/Storm? Well, bunches of them originated in Saudi. We would hook
the RS-232 (I/O) of a FAX machine to a Harris MIL-STD-188-110c modem run
the 39 tone mode and FAX them out to the troops at 2400 BPS using only
100 watts (AN/URC-119 HF system by Harris). Data input was thru the
aux. audio jack (a DB-9) and running USB. We would pass a page in just
a couple of minutes.

Ok, so how did the Patriot missile know where to "point" in the sky to
lock onto a Skud? Easy, the AWACS caught the ground launch, snapped a
video picture from the radar and sent the picture, with Lat/Long data,
via HF radio and a MIL-STD-188-110c modem down the the Harris HF radio
and MIL-STD modem in the Patriot missile battery and the battery pre-
positioned its radar antennas and missile tubes in the correct part of
the sky to intercept the Skud. And all this time you thought the
Patriot missile was doing it all. Yep AWACS to Patriot in less than a
minute. Did I mention digital encrypted voice on HF? Guess I'd better
not on that one.

Hello from Levenworth.

73,

37. Achieving regulation by bandwidth

From: BobH@eworld.com
Date: Wed, 21 Dec 94 04:23:15 PST
To: hfsig@tapr.org
Message-ID: <9412210423.tn19172@eworld.com>
Rendered body type: text/plain

I'm glad someone mentioned the appropriateness of a modulation technique for "noisy, interference-prone" channels. Narrow-band co-channel interference (QRM) is getting my attention as the missing sun spots force more hf users into competition on lower and lower and fewer and fewer frequencies.

Analyze your modulation scheme's BER in white noise and then analyze it again this time with a +30dB 850 shift 75 baud binary FSK signal in the middle of it.

The commercial single tone modems waveshape spread to 3 khz and then use adaptive narrow band interference suppressers to notch out several jammers on the channel. The only energy to "shine" on the detector is the wanted signal.

QRM is a good argument for relaxing the bandwidth limits for new digital modes (in what at first seems to be a paradox but really isn't as others on this SIG and elsewhere have pointed out). We can co-exist with current user's: We notch them out; they power through us. Or our signal processing gain spreads them and de-spreads us, something like that.

If we don't have the MIPS and the algorithms for the STANAG and the MIL-STD adaptively equalized single tone hf modems, I saw an alternative: "A chirp modem incorporating interference excision" IEE Proceedings Vol. 139, Aug 1992 .

They chirp from 300-3000 Hz using a TMS320C25 and "recent work has increased the data rate to 600 bits/s". Looks easy, fun, robust and illegal!

Bob

38. Re: [HFSIG:149] Re: HF Modulation Modes

From: "Johan Forrer
Date: Wed, 21 Dec 1994 08:43:35 PST8PDT
To: hfsig@tapr.org
Message-ID: <B68A31F1F55@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Alan Bloom, N1AL wrote:

>On the receiving end, I presume you would just run a DFT on the
>incoming sampled data which would output the frequency and phase
>of each carrier. It is effectively coherent detection, since
>the phase is known.

That is good news Alan. After your post, I spent last night digging out all the stuff that I had on the MIL-STD-188 39-tone parallel modems, and what a surprise - now that I know a little better what to look for.

The 39-tone system is, by definition, what you called ODFM. It comprizes 39 tones, spaced to obtain orthogonal signaling, with each tone TDPSK (4 phase) modulated (I don't quite understand yet what this means? Anyone pse). Although very few manufacturers actually tell you how they do their demodulation, I found one reference to it: 128 point FFT! All these systems occupy a typical voice bandwidth channel and even when it operates at lower bit rates, never uses less bandwidth, but uses redundancy instead. The figures for BER for these 39-tone modems, as specified minimum performance specs (from MIL-STD-188-110 document), requires:
1.8e^-4 at 30 dB S/N for 2400 bps, 2.7e^-6 also at 30 dB S/N for 1200 bps. At 75 bps 1.0e^-6 for 8 dB S/N. I think wwill agree that this is not bad. Keep in mind that all these BER merits relies heavily on coding, i.e. block or convolutional combined with interleaving. All the specs that I looked at also includes some form of tracking for doppler, i.e. +/- 75Hz, at a rate of 3.5 Hz/second. One small tidbit was that one manufacturer actually disclosed that an M68030 was used as a controller in conjunction with two M56002's.

Does anyone have further references to such ODFM modulation schemes? I would be very interested to hear more.

Thanks for all the excellent contributions - we are making good progress.

73's

Johan, KC7WW

39. Re: HF Modulation Modes

From: Larry Gadallah <larry@owrlakh.unbc.edu>
Date: Wed, 21 Dec 1994 09:15:19 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199412211715.JAA08836@owrlakh.unbc.edu>
Rendered body type: text/plain

>
> In all the discussion about the best modulation format for HF, I'm
> surprised nobody has mentioned OFDM. Orthogonal frequency-division
> multiplex is proposed for Digital Audio Broadcasting and digital TV
> in Europe, although it is not so well known in the US.
>
> The idea is to split the signal up into a large number (like 128 or
> 1024) equally-spaced carriers, each modulated at the same symbol rate.
> I believe the DAB standard proposes that each carrier be modulated
> with QPSK, but you can use any modulation in which each symbol can be
> represented by a phase and amplitude. That includes the many forms of
> QAM, PSK, PI/4 DQPSK, on-off keying, whatever.
>

Isn't this what Telebit does for its Packetized Ensemble Protocol (PEP)
used for long haul high-speed landline modem connections? Some of my
colleagues and I have occasionally wondered about using a couple of
Telebits for HF modems, despite the obvious legal problems.

I have also considered the idea that OFDM is a coarse grained form of
spread spectrum, albeit with little spreading. Wouldn't it be nice
to be able to spread a 1 kbps signal over a 3 khz bandwidth and thus
provide some protection against interference and fades? How about
feeding a 1 kbps data stream into a DSP and producing a 300 to 3300
Hz DS spread spectrum signal that could be piped directly into the
modulator of an HF rig. I guess there could be problems with modulator
non-linearity.

73,

--

Larry Gadallah VE4TCP/VE6VQ Prince George, BC
Internet: larry@owrlakh.unbc.edu TPC: (604) 964-1140
 ICBM: 53:51'38.4"N 122:44'25.8"W

40. Re: [HFSIG:153] Re: HF Modulation Modes

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Wed, 21 Dec 94 20:15:23 MET
To: hfsig@tapr.org
Message-ID: <9412211915.AA21592@ife>
Rendered body type: text/plain

> ... One small tidbit was that one manufacturer
> actually disclosed that an M68030 was used as a controller in conjunction
> with two M56002's.

Recently, I had the chance to have a view into a pair of pre-production MIL-STD-188
parallel- (and serial-) tone HF-modems (the kind of olive-green painted brand and price-tag).
The job is done by two ADSP-2100 16-bit fixed point DSPs from Analog Devices.
This job can probably done soon by a single DSP like a 80 or 100 MHz TMS320C50 or similar.

73s,

Rolf

--

--- * * * ---

Rolf Sommerhalder	home address:
Swiss Federal Institute of Technology (ETH)	c/o Rindlisbacher
Electronics Laboratory ETZ H60.1	Wermatswilerstrasse 96
Gloriastrasse 35	8610 Uster / Switzerland
8092 Zurich / Switzerland	
Voice +41 1 632 76 40 (or 632 51 53)	Voice +41 1 941 59 28
Fax +41 1 632 12 10	AX.25 HB9CWP @ HB9OS
E-Mail rolf.sommerhalder@ife.ee.ethz.ch	Swiss Disaster Relief HBK81

41. Re: HF Modulation Modes

From: BLOOM_ALAN/HP5300_R0@opnmail3.corp.hp.com
Date: Wed, 21 Dec 94 11:48:00 -0800
To: hfsig@tapr.org
Message-ID: <d0eqY8Z00000000000@MHS>
Rendered body type: text/plain

Walt@home.2.nite said:

>Gee Alan, kinda sounds like the 16 and 39 parallel tone modem formats in
>MIS-STD-188-110c. ...

Kinda. Although with only 16 or 39 tones, using a FFT/IFFT is not very efficient, since you don't have 2^N tones. With fewer tones, you need a higher symbol rate to get a desired bit rate, which would require more DSP horsepower. As Bob said:

BobH said:

>If we don't have the MIPS and the algorithms for the STANAG and the
>MIL-STD adaptively equalized single tone hf modems...

The advantage of OFDM is the computational efficiency due to using the FFT. Equalization is easy: Just adjust the amplitude and phase of each of the carriers in the frequency domain before doing the IDFT.

Larry Gadallah said:

>I have also considered the idea that OFDM is a coarse grained form of
>spread spectrum, albeit with little spreading. ...

It's not really spread, since the total bandwidth is about the same as a single carrier modulated with the same bit rate.

By the way, on the subject of bandwidth, there is no need for pre-modulation filtering of the transmitted signal (other than the standard DAC anti-aliasing filter). That's because each carrier has such a low baud rate that even with unfiltered square waveforms the $\sin x/x$ frequency rolloff is fast enough to give an almost-square frequency spectrum for the composite signal.

Alan

42. Re: [HFSIG:141] Re: Achieving regulation by bandwidth

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Wed, 21 Dec 1994 15:29:12 -0800
To: FORRERJ@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199412212329.PAA05236@unix.ka9q.ampr.org>
Rendered body type: text/plain

Johan,

This stuff is well covered in communications theory textbooks, but here's a shot.

Entropy is a measure of information content. It's also a measure of uncertainty. Receiving a bit that you always know will be a "1" or a "0" carries no real information; its entropy is 0 bits. A bit that is (to you) completely unpredictable and has a 50-50 chance of being either a 1 or a 0 carries 1 bit of entropy. A bit that is skewed, i.e., has some uncertainty to it but can be predicted with better than chance odds has entropy somewhere between 0 and 1 bit.

Compression algorithms increase entropy in a data stream by removing redundancy. An ideal compression algorithm produces output with 1 bit of entropy per compressed data bit.

Channel capacity on an additive white gaussian noise channel is defined by Shannon's famous formula

$$C = B \log_2(1 + S/N)$$

where C = channel capacity in bits/sec ("real" bits, with 1 bit entropy each)
B = channel bandwidth in Hz
S = received signal power in watts
N = received noise power in watts

The important thing to note about the Shannon equation is the fundamental tradeoff between bandwidth and S/N. The wider the bandwidth, the lower the S/N required to maintain a constant capacity. And the narrower the bandwidth, the higher the S/N required.

Since N also depends on bandwidth it's usually more instructive to consider E_b/N_0 instead of S/N:

$$E_b/N_0 = (S/R) / (N/B) = S/N * B/R$$

where E_b = received energy per bit in joules
 N_0 = received noise spectral density in watts/Hz.
 R = data rate in bits/sec

Note that E_b/N_0 is independent of signal bandwidth. E_b/N_0 is still dimensionless (just like S/N) since watts/Hz also has units of joules. It is equal to S/N only for the case where the modulation bandwidth in Hz equals the data rate in bits/sec. E_b/N_0 is now universally used in describing the performance of a modem against white noise.

If you substitute this into Shannon's formula and solve for E_b/N_0 you get

$$E_b/N_0 = B/R * (2^{(R/B)} - 1)$$

This tells you the absolute minimum E_b/N_0 that any modem must have when operating at a specified data rate R and bandwidth B . Real modems will of course require more. In the limit of $B/R = \text{infinity}$, i.e., infinite bandwidth for a finite data rate, $E_b/N_0 = -1.6$ dB, the famous Shannon limit. Note, though, that this applies ONLY to infinite bandwidth. For finite values of R/B , the required E_b/N_0 is higher. E.g., for R/B of unity, the minimum E_b/N_0 is 0 dB. For an R/B of about 4 bit/sec/Hz, the minimum E_b/N_0 is about +6 dB. If you make R/B less than 1, eg, by adding FEC, the minimum required E_b/N_0 drops below 0 dB. For a rate 1/2 code, this is about -.8 dB. Real, implementable codes at this rate can get down to about +2.5 dB; I've done this with soft decision sequential decoding.

Note that a "bit" represented by E_b is a real user data bit, NOT a coded channel symbol. That's another reason to use E_b/N_0 : it takes FEC into account. FEC provides a real power gain in that I can use fewer transmitter watts to send a given number of user data bits per second. But there's no free lunch; my transmitter power savings come at the expense of additional bandwidth. FEC simply allows me to play with the terms in the Shannon equation and to come more closely to its limits; it doesn't allow me to violate it.

Power efficiency has traditionally been considered important only in situations like deep space communications where transmitter power is an absolute premium. But it's now getting quite a bit of attention even where RF power is relatively cheap: the shared terrestrial channel. That's because a modulation method that requires relatively low E_b/N_0 is not only more tolerant of interference (considering interference to be part of the noise) but it also generates less interference to others since it needs less power to communicate.

This is a win-win approach much like reducing weight in a car: the lighter you make the car frame, the lighter the engine you need to move it, which means an even lighter frame to carry the weight of the engine, which means an even lighter engine, and so on.

It turns out that at least in cellular mobile communications, the savings from reduced interference and increased interference tolerance more than make up for the extra bandwidth required. This is the basis of the CDMA spread spectrum stuff we've been developing at Qualcomm.

I'm convinced that same principles apply to *any* shared, interference limited RF environment, including HF. John Costas said as much when he wrote his famous paper, "Poisson, Shannon and the Radio Amateur" that appeared in the Proceedings of the IRE, December 1959. (It has been reprinted in many places, such as the IEEE book "Multiple Access Communications", edited by Norman Abramson, ISBN 0-87942-292-0.)

Phil

43. Re: [HFSIG:145] Re: HF Modulation Modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 22 Dec 1994 01:59:58 -0800
To: FORRERJ@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199412220959.BAA05620@unix.ka9q.ampr.org>
Rendered body type: text/plain

>This is interesting - one question: did we overcome the need for
>coherent detection, or does it matter?

>From briefly browsing the MIL specs, it looks like they dedicate one
of the carriers to serve as an unmodulated phase and frequency
reference for the others.

Qualcomm CDMA does something similar on the forward (base->mobile)
link. There's a spread but otherwise unmodulated component called the
pilot that is always present. It's several dB stronger than the
traffic channels. The mobiles use it as a timing, phasing and
frequency reference for everything else they do. This works well in
our system because a single pilot can be shared by all of the channels
in the system. The reverse link has no pilot, so we use a form of
64-ary orthogonal modulation instead to make up most of the loss from
noncoherent demodulation.

Phil

44. Re: [HFSIG:149] Re: HF Modulation Modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 22 Dec 1994 02:18:15 -0800
To: BLOOM_ALAN/HP5300_R0@opnmail3.corp.hp.com
Cc: hfsig@tapr.org
Message-ID: <199412221018.CAA05738@unix.ka9q.ampr.org>
Rendered body type: text/plain

>On the receiving end, I presume you would just run a DFT on the
>incoming sampled data which would output the frequency and phase
>of each carrier. It is effectively coherent detection, since
>the phase is known.

Not necessarily. Just because you can measure the received phase
doesn't imply you know what phase was sent. But if you dedicated one
of the channels to be an unmodulated phase reference, then you could
probably use this to determine the transmitted phase in the other
channels, assuming that the channel isn't too dispersive.

Phil

45. Re: [HFSIG:151] Re: HF Modulation Modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 22 Dec 1994 02:49:23 -0800
To: k5yfw@k5yfw.ampr.org
Cc: hfsig@tapr.org
Message-ID: <199412221049.CAA05769@unix.ka9q.ampr.org>
Rendered body type: text/plain

Hmm, MIL-STD-188-110C...is that anything like MIL-STD-188-110A? I just
ran off a bunch of copies of the latter for the people who requested
them, and now I wonder if I didn't have the latest revision. The date
on the -110A version is 30 September 1991, which superseded -110 dated
15 November 1980.

If -110A didn't become active until late 1991, after the Gulf War was
over, then how could you have used -110C modems? From looking at the
standard it sure does *seem* like the same thing; there's a 16 tone
mode and a 39 tone mode.

By the way, for anyone interested in actually implementing this spec,
I note that it specifies a rate 1/2 K=7 convolutional FEC. Several
months ago I wrote and debugged a fast soft-decision Viterbi decoder
for this particular code as it's a widely used standard (e.g., the
Voyager spacecraft).

After considerable cycle bumbing my decoder runs at 41 kb/s on a 66
Mhz 486 (32-bit protected mode, GCC 2.5.8 with full optimization).
Should be plenty fast for this application. I'd be happy to make it
available.

Phil

46. Re: [HFSIG:151] Re: HF Modulation Modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 22 Dec 1994 03:15:12 -0800
To: k5yfw@k5yfw.ampr.org
Cc: hfsig@tapr.org
Message-ID: <199412221115.DAA05783@unix.ka9q.ampr.org>
Rendered body type: text/plain

>Ok, so how did the Patriot missile know where to "point" in the sky to
>lock onto a Skud? Easy, the AWACS caught the ground launch, snapped a
>video picture from the radar and sent the picture, with Lat/Long data,
>via HF radio and a MIL-STD-188-110c modem down the the Harris HF radio
>and MIL-STD modem in the Patriot missile battery and the battery pre-

Hmm, considering that the more sober studies made after the war
concluded that there was no hard evidence that even ONE SCUD missile
had been successfully engaged and destroyed, perhaps we should
consider a different HF modem technology?

:-)

More seriously, I did get a chance to leaf through the Ralphs book
this afternoon in the UCSD library. I hadn't renewed my library card,
so I had to get the gist of it in the 45 min remaining before the
library closed. (Bad planning, I know). I also noted the comparisons
between the Piccolo-style MFSK modems used by the British foreign service
and the multicarrier PSK systems that seem to be favored by the US
military. The Eb/N0 curves were significantly better for Piccolo. The
discussion commented that the multicarrier PSK systems did require much
better channel conditions.

This matches my understanding of the problem. Now it is true that real
HF channels are not always as bad as the pure Rayleigh model would
suggest. The fading process is usually bandlimited, limiting the fade
rate, and there are usually only a small number (often 2) of multipath
components. In this situation you *can* get away with phase
modulation.

But if you really want to get the most out of a really lousy channel
with the least power, I think the author is right - MFSK with
noncoherent detection is the way to go. Another part of the book
described some interference tests they did. One involved running
Piccolo signals "underneath" BBC program material. The Piccolo signal
could still be copied even when it was so weak as to be only a
slightly noticeable background behind the audio program.

Phil

47. HF Modulation Modes

From: BobH@eworld.com
Date: Thu, 22 Dec 94 04:17:50 PST
To: hfsig@tapr.org
Message-ID: <9412220417.tn32612@eworld.com>
Rendered body type: text/plain

I just got my "QEX" for December:
STANAG 4285 kicks butt!

Which leads to one question (of many)
to which I'm sure no simple answer
exists:

Which code is "better" for the hf channel:
> A convolutional code with interleaver.
or
> A R-S block code with soft decision.

????

Or is that even the right question?

If we only had a channel simulator.

bob wm6h

48. Re: [HFSIG:101] Ideas for HF modulation and coding

From: "Johan Forrer"
Date: Thu, 22 Dec 1994 09:03:49 PST8PDT

FL" <FORRERJ@frl.orst.edu>

To: hfsig@tapr.org
Message-ID: <B80FA331EF6@frl.orst.edu>
Rendered body type: text/plain

Phil,

Thanks much for the brief note on Shannon - it always is a good reality check to keep those principles in mind.

OK - I am ready to start playing with some of these great ideas - perhaps we can start looking into some of the practical details now. It would not be difficult to simulate the multi-tone signals in a C program, and then play with the different ideas on demodulation.

It appears there are two ideas put forward:

1) (ODFM) n-parallel carriers with orthogonal signaling. All carriers are always present. Demodulation is performed by FFT's. It is obvious that the choice of the FFT size and sample rate have to just right otherwise we will end up with an odd distribution of our carriers over the frequency/phase bins. After this choice have been made, how do I use this I/Q recovered data? I have some wild ideas, but would like to obtain further input first.

2) (Phil's method based on Welch functions) n-parallel tones with orthogonal signaling. Only a small subset of carriers are present at a time - the different subset is selected from a Hamadard matrix. In a sense, Piccolo may be considered a special case of this idea, where only one carrier is sent at a time from a set of m-carriers. Phil mentioned that FHT (is that for fast hadamard transform?) is used to perform demodulation. How about telling us a bit more about how to implement this one Phil.

73's

Johan, KC7WW

49. Which Modem

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Thu, 22 Dec 94 11:01:21 -0600
To: hfsig@tapr.org
Cc: tbruns@datarace.com
Message-ID: <m0rKqtF-00024cC@sacdm10.kelly.af.mil>
Rendered body type: text/plain

The question has been ask "which modem?"

I think the answer is, we don't know at this point in time.

While members of this group are gathering information about different modems and techniques, they are also gathering information about a how to create an HF channel simulator. Once a simulator is developed, we can evaluate different modems and techniques. The group can then begin to evaluate different modems and modem techniques.

I wish we were already to release code for a soundcard that would make 2400 BPS possible on HF. But, most/all of us have to put beans on the table for ourselves and/or family. I keep telling myself this *is* a hobby.

When it comes time to release code for on-the-air testing, and before, we are going to need an informed member base to carry our torch to the rest of the ham world...this isn't going to be easy and we will have a great amount of resistance. But, we must be ready, informed and ready to fight.

One thing to keep in mind is that *when* we are successful, we will be giving the ham radio world a wonderful new "toy" and make robust, high speed data transmissions on HF available at an affordable price to the entire world.

73 & Happy Holidays,

Walt/K5YFW

50. Re: [HFSIG:161] Re: HF Modulation Modes

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Thu, 22 Dec 94 11:38:03 -0600
To: hfsig@tapr.org

Cc: tbruns@datarace.com
Message-ID: <m0rKrSm-0002ANC@sacdm10.kelly.af.mil>
Rendered body type: text/plain

In your message of 22 Dec 1994 at 0520 CST, you write:
> >Ok, so how did the Patriot missile know where to "point" in the sky to
> >lock onto a SCUD? Easy, the AWACS caught the ground launch,
snapped a
> >video picture from the radar and sent the picture, with Lat/Long data,
> >via HF radio and a MIL-STD-188-110c modem down the the Harris HF radio
> >and MIL-STD modem in the Patriot missile battery and the battery pre-
>
> Hmm, considering that the more sober studies made after the war
> concluded that there was no hard evidence that even ONE SCUD missile
> had been successfully engaged and destroyed, perhaps we should
> consider a different HF modem technology?
>
> :-)

The video taken by my troops from atop of the villas in Eskan Village just outside of the Saudi Capital (50,000 + U.S. troops lived in Eskan Village) indicated that there were direct hits. Whether they destroyed, deflected or made a difference in the targeting is an opinion.

The data provided by the HF to the Patriot sites only helped the site point their radar antennas in the right part of the sky, it had nothing to do with the radar acquisition/lockon of the SCUD or the capability of the missile system destroying the target. A Patriot Missile Battery cannot "look" at the entire "sky" in front of it, just one quadrant.

>
> More seriously, I did get a chance to leaf through the Ralphs book
> this afternoon in the UCSD library. I hadn't renewed my library card,
> so I had to get the gist of it in the 45 min remaining before the
> library closed. (Bad planning, I know). I also noted the comparisons
> between the Piccolo-style MFSK modems used by the British foreign service
> and the multicarrier PSK systems that seem to be favored by the US
> military. The Eb/N0 curves were significantly better for Piccolo. The
> discussion commented that the multicarrier PSK systems did require much
> better channel conditions.
>
> This matches my understanding of the problem. Now it is true that real
> HF channels are not always as bad as the pure Rayleigh model would
> suggest. The fading process is usually bandlimited, limiting the fade
> rate, and there are usually only a small number (often 2) of multipath
> components. In this situation you *can* get away with phase
> modulation.
>
> But if you really want to get the most out of a really lousy channel
> with the least power, I think the author is right - MFSK with
> noncoherent detection is the way to go. Another part of the book
> described some interference tests they did. One involved running
> Piccolo signals "underneath" BBC program material. The Piccolo signal
> could still be copied even when it was so weak as to be only a
> slightly noticeable background behind the audio program.
>

Do we want to design a modem for the worse-case scenario or can we live with one that can get data thru on the worst channel part of the time. I have a pick-up without 4 wheel drive...there are at times places I can't drive...so I don't drive there until conditions are better. There are places where I can drive without 4 wheel drive that my Geo Metro can never go. Should I have 4 wheel drive? Do we need a modem that can get thru even the worst conditions or can we settle for something less. Are we purist or realists?

Walt (dba k5yfw)

51. Re: [HFSIG:160] Re: HF Modulation Modes

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Thu, 22 Dec 94 12:17:44 -0600
To: hfsig@tapr.org
Message-ID: <m0rKs5B-000295C@sacdm10.kelly.af.mil>
Rendered body type: text/plain

Phil,

In your message of 22 Dec 1994 at 0456 CST, you write:

> Hmm, MIL-STD-188-110C...is that anything like MIL-STD-188-110A? I just
> ran off a bunch of copies of the latter for the people who requested
> them, and now I wonder if I didn't have the latest revision. The date
> on the -110A version is 30 September 1991, which superseded -110 dated
> 15 November 1980.
>
> If -110A didn't become active until late 1991, after the Gulf War was
> over, then how could you have used -110C modems? From looking at the
> Standard it sure does *seem* like the same thing; there's a 16 tone
> mode and a 39 tone mode.

Make that 110Xb, The appendix(?) for HF modems in the copy I have shows "b" for the 39 tone modem and "c" for the serial tone Harris modem. I don't have the complete MIL-STD, just the part on the modems...and that was given to me by the Harris RF Comm. 1Div.

I'm glad someone is keeping me straight and honest, my XYL, WB5WXY, has failed at that task for the last 30 years.

Walt

52. Frederick Electronics Modem

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Thu, 22 Dec 94 12:46:27 -0600
To: hfsig@tapr.org
Cc: tbruns@datarace.com
Message-ID: <m0rKsWx-000295C@sacdm10.kelly.af.mil>
Rendered body type: text/plain

High Performance Data Modem

SFA DataComm, Inc.
Frederick Electronics
7450 New Technology Way,
P.O. Box 502
Frederick, MD 21701
(301) 662-5901

Model 1102 NSN 5895-66-136-4191 (\$7,500 U.S.)

Capable of Tx and RX up to 2400 BPS in a standard 3 KHz ssb HF radio channel...meets Mil. Standard 188-110A

3.47"H x 17"W x 16"D (19" Rack mounting) Approx. 12 lbs

....Convolutional coding with soft decision Viterbi decoding....

....Interleaving...

...User friendly interface...movement between windows...

Additional planned capabilities:

16 tone DPSK per MIL-STD-188-110a
39 tone DPSK per MIL-STD-188-110a
NATO mode IAW STANAG 4285 (3600 BPS)
16 channel VFT mode (R.39)

53. Fw: Oops, I lied Again

From: k5yfw (WALT DUBOSE - K5YFW)
Date: Thu, 22 Dec 94 12:28:47 -0600
To: hfsig
Rendered body type: text/plain

Oops, I lied again. The copy of the part MIL-STD-188-110 that I have says:

MIL-STD-188-110
NOTICE 2
4 November 1988

APPENDIX B, 16-TONE DIFFERENTIAL PHASE-SHIFT KEYING (DPSK) MODE
APPENDIX C, 39-TONE MODE

Walt

54. Re: [HFSIG:161] Re: HF Modulation Modes

From: jmorriso@bogomips.ee.ubc.ca (John Paul Morrison)
Date: Fri, 23 Dec 1994 10:43:19 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0rLExU-0004giC@bogomips.ee.ubc.ca>
Rendered body type: text/plain

> But if you really want to get the most out of a really lousy channel
> with the least power, I think the author is right - MFSK with
> noncoherent detection is the way to go. Another part of the book
> described some interference tests they did. One involved running
> Piccolo signals "underneath" BBC program material. The Piccolo signal
> could still be copied even when it was so weak as to be only a
> slightly noticeable background behind the audio program.

This isn't related to HF, but: would some modulation like Piccolo (I think I missed the description for that) be applicable to sending a data stream over a 2m or 70cm narrow-band FM voice radio? And preferably through a voice repeater as well. Speed would be about 50 bps or so, just enough to send your callsign or a very short message when you key up the radio.

>
> Phil
>

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
jmorriso@bogomips.ee.ubc.ca ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org

55. HF modulation

From: Johan Forrer <70730.3472@compuserve.com>
Date: 26 Dec 94 18:25:39 EST
To: HFSIG <HFSIG@tapr.org>
Message-ID: <941226232538_70730.3472_CHK57-1@CompuServe.COM>
Rendered body type: text/plain

Hi,

As a way to learn more about how OFDM works for the 39-tone MIL-STD-188 parallel modem, I wrote a C program that simulates 39 parallel tones and combines them into a single output signal. The program then continues by modulating (DQPSK) each of these tones, using a simple binary pattern, i.e., 00 -> 01 -> 10 -> 11 (remember these are di-bits). This gives an effective raw rate of $2 \times 39 \times 56.25 = 4387.5$ bps (56.25 baud for each of the 39 channels) using a 3 kHz voice channel. Although I have not explored further aspects as yet, several transmission speeds ranging from 75 - 2400 bps are possible by using some of the channels for redundancy. Error-correction coding is further included that trades off between these bit rate and robustness.

The program then continues the demodulation process by capturing 128 data points and performing an FFT. This produces 64 positive frequencies of which only 39 are used - the remainder are disregarded or treated as guard bands - as would be required when such a system are to be used with a voice-grade IF filter. Shown in Table 1, below, are the recovered I and Q data

for four channels, i.e., the first four channels of 39. It is evident that, with proper bit synchronization (which is fairly well documented for the MIL-STD modems), things seem to work out remarkably well.

I also did capture the simulated output waveform and was quite concerned after inspecting the plot - the presence of large spikes of significant magnitude (in some cases, some 10-15 dB over the RMS value). This situation will certainly introduce data errors if not addressed. It should be stressed that such a signal will sound like a noise burst - it has little or no melody associated with it, except for the bit synchronizing pre-amble.

It is in my opinion, that after this initial analysis, that such a scheme may be quite doable on our present hardware. I have no doubt that, if signal synthesis and fast Fourier transform are performed on the DSP side, there should be plenty of computing power in a modest PC to run the remainder of such a modem, even considering the computing demands for error coding.

Hope you all enjoyed the holiday season. Trust that 1995 will hold the best in health peace for you.

73's

Johan, KC7WW

Table 1.

An evaluation of the MIL-STD-188 modem using OFDM.

BAUD #	I	Q	Freq	Real	Imag	<- appears swapped
1	0	0		675.00	45.25	-45.25
2	0	1		675.00	-45.26	45.24
3	1	0		675.00	45.28	-45.23
4	1	1		675.00	-45.31	45.20
1	0	1		731.25	-45.25	45.25
2	1	0		731.25	45.26	-45.24
3	1	1		731.25	-45.22	-45.27
4	0	0		731.25	45.23	45.24
1	1	0		787.50	45.25	-45.25
2	1	1		787.50	-45.24	-45.26
3	0	0		787.50	45.24	45.28
4	0	1		787.50	-45.30	45.22
1	1	1		843.75	-45.25	-45.25
2	0	0		843.75	45.24	45.26
3	0	1		843.75	-45.25	45.23
4	1	0		843.75	45.20	-45.31

KC7WW, Dec 26, 1994

56. copies of MIL-STD-188-110A

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 26 Dec 1994 17:01:51 -0800
To: hfsig@tapr.org
Message-ID: <199412270101.RAA18816@servo.qualcomm.com>
Rendered body type: text/plain

As I mentioned a week or so ago, I've run off four paper copies of MIL-STD-188-110A, dated 30 September 1991.

I've also printed MIL-STD-203-1A, "Interoperability and Performance Standards for Tactical Digital Information Link (TADIL) A", 8 January 1988, and MIL-STD-188-203-3, "Subsystem Design and Engineering Standards for Tactical Digital Information Link (TADIL) C", 5 October 1983. These two standards seem to discuss earlier multitone modems, perhaps related to 110A.

All standards are completely unclassified and marked "Approved for Public Release; distribution is unlimited."

I'm sending a copy to Frode, LA2RL. Any other takers? If I get more than another 3, I can send the laserprinter originals to Greg and he can run them off with TAPR resources, right Greg? :-)

Phil

57. Re: [HFSIG:160] Re: HF Modulation Modes

From: Larry Gadallah <larry@owrlakh.unbc.edu>
Date: Sun, 25 Dec 1994 16:54:27 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@unix.ka9q.ampr.org
Message-ID: <199412260054.QAA00731@owrlakh.unbc.edu>
Rendered body type: text/plain

> By the way, for anyone interested in actually implementing this spec,
> I note that it specifies a rate 1/2 K=7 convolutional FEC. Several
> months ago I wrote and debugged a fast soft-decision Viterbi decoder
> for this particular code as it's a widely used standard (e.g., the
> Voyager spacecraft).
>

> After considerable cycle bumbing my decoder runs at 41 kb/s on a 66
> Mhz 486 (32-bit protected mode, GCC 2.5.8 with full optimization).
> Should be plenty fast for this application. I'd be happy to make it
> available.
>
> Phil

I for one would be interested.

Thanks and 73,

--

Larry Gadallah Prince George, BC
Internet: larry@owrlakh.unbc.edu TPC: (604) 964-1140
VE4TCP/VE6VQ ICBM: 53:51'38.4"N 122:44'25.8"W

January 1995

Original source file: hfsig.9501

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1995-01/TAPR-HF_1995-01.txt

1. HF Modulation

From: "Johan Forrer
Date: Tue, 3 Jan 1995 08:50:59 PST8PDT
To: HFSIG@tapr.org
Message-ID: <CA0CBB055C2@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi All,

Trust you had a good holiday season - best wishes for 1995.

Apparently there were some problems with the listserver - hope it is working now.

Bob asked an interesting question: whether soft decision decoding with convolutional coding was as effective as linear block coding with interleaving. There is an interesting, though rather oldish, paper about this that you may find interesting:

"Effective Application of Forward-Acting Error-Control Coding to Multichannel HF Data Modems" by Pierce et. al. IEEE Trans. on COMM, Vol COM-18 (4), August 1970. pp 281-294.

This paper compares early multi-tone parallel modems over HF paths and looks at how different phase and frequency shift modulation formats perform in conjunction with various linear block, as well as, convolutional codes. It appears that some block codes (i.e. (23,12)) code with interleaving does as well as a convolutional code with the appropriate constraint length (very large), however, it is not a simple matter to engineer the coding scheme without regard to the modulation scheme - these things should be engineered to work together and not separate. This comparison was by live experiments over transcontinental as well as transatlantic HF paths, and is about as old as the work done by Watterson on the HF channel simulator.

I still am looking for further information on the Fast Hadamard Transform (or the Fast Walsh Transform). If anyone can shed some light on this, it would be greatly appreciated.

73's

Johan Forrer, KC7WW

2. Re: [HFSIG:170] HF modulation

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Tue, 03 Jan 95 20:34:06 CST
To: hfsig@tapr.org
Message-ID: <2428@k5yfw.ampr.org>
Rendered body type: text/plain

Hi Gang,

Wanna see video of a 53 year old ham sledding down a 1000 ft hill on a plastic sled? Wanna see the old feller go bump...wanna see the old feller's 2 1/2 year old granddaughter say "Owie Pappa?" Sun Valley (Ketchum?) will never be the same again. And no one answered my call on the 147.18 MHz Ketchem repeater.

Boooooo!!!!

In message <941226232538_70730.3472_CHK57-1@CompuServe.COM> Johan writes:
[stuff deleted]

> I also did capture the simulated output waveform and was quite concerned
> after inspecting the plot - the presence of large spikes of significant
> magnitude (in some cases, some 10-15 dB over the RMS value). This
> situation will certainly introduce data errors if not addressed. It should be
> stressed
> that such a signal will sound like a noise burst - it has little or no melody
> associated with
> it, except for the bit synchronizing pre-amble.

I heard a couple of Harris engineers talking about the problem once...I think they fixed it in there MIL-STD-188-110 modems 39

tone modem.

Boy does the signal ever sound like noise...as you tune across the signal the S meter will rise and fall and you will think you are just tuning a noisy frequency.

You need a receiver capable of 10 Hz resolution to receive this modulation and if you're receiver (transceiver) dial/readout doesn't show 10 Hz steps, you will need to have a tuning indicator built into the software.

>
> It is my opinion, that after this initial analysis, that such a scheme may be
> quite doable on our present hardware. I have no doubt that, if signal synthesis
> and fast
> Fourier transform are performed on the DSP side, there should be plenty of
> computing power in a modest PC to run the remainder of such a modem, even
> considering the computing demands for error coding.

Sounds good to me.

>
> Hope you all enjoyed the holiday season. Trust that 1995 will hold
> the best in health peace for you.

Yep, my holidays were fun fun fun. Spoiled my granddaughter again just to get back at my son.

73 & CUL,

Walt@home.2.nite

3. subscribe

From: dlmill@mtu.edu (Dan Miller)
Date: Wed, 4 Jan 1995 10:36:17 -0500
To: hfsig@tapr.org
Message-ID: <199501041536.KAA15878@mtu.edu>
Rendered body type: text/plain

Could someone please email me how to subscribe to this list?
Dan dlmill@mtu.edu

4. Re: [HFSIG:175] subscribe

From: Stephen Hall <shall@rain.org>
Date: Wed, 4 Jan 1995 09:05:57 -0800 (PST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SUN.3.90.950104090343.17248A-100000@coyote.rain.org>
Rendered body type: text/plain

As I've forgotten, I too would like this posted. I have some friend that would like to subscribe additionally.

On Wed, 4 Jan 1995, Dan Miller wrote:

> Could someone please email me how to subscribe to this list?
> Dan dlmill@mtu.edu
>
>
>

5. Re: [HFSIG:176] Re: subscribe

From: "Johan Forrer
Date: Wed, 4 Jan 1995 09:13:17 PST8PDT
To: hfsig@tapr.org
Message-ID: <CB92BCD39FA@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi,

Hope this helps.

Greetings from TAPR!

TAPR can be reached via the Internet. The Automated Information Server that TAPR provides allows anyone to request information on TAPR, products, newsletters, and lots of other files. To find out more about this service, send an e-mail message to listserv@tapr.org with the subject line "Request" and the following text in the body of the message:

```
help                (for a brief set of instructions)
index -all          (for a list of all file by topic area)
list                (for a list of TAPR Mail Groups)
get tapr taprinfo.txt (or info on TAPR)
```

In the above message example, "help" retrieves a brief set of instructions for info, "index -all" retrieves a list of available files by topic and "taprinfo.txt" retrieves a text file containing information on TAPR and what TAPR is all about. If you want to retrieve several text files with one message, use a separate line for each "get filename" request.

General Information:

- * Ever heard of a TNC ?
- * Interested in general packet radio information?
- * Interested in high-speed packet operations?
- * Interested in other types of digital communications?
- * Interested in experimenting or building kits?
- * Interested in keeping up-to-date on national digital and packet issues?

Then you might be interested in TAPR!

Goals:

- % Support Research and Development efforts in the area of amateur digital communications
- % Disseminate information on packet and digital communications
- % Provide affordable and useful kits for experimenters and hobbyists
- % Pursue and help advance the amateur art of communications through publications, meetings, and standards

History:

If you have used a packet radio TNC, then you are already a part of TAPR history.

The TNC (Terminal Node Controller) project grew from a discussion in October of 1981 at a meeting of the Tucson Chapter of the IEEE Computer Society. A week later, six of the attendees gathered and discussed the feasibility of developing a Terminal Node Controller that would be complete and available to amateurs at a modest cost. This was the genesis of TAPR.

On June 26th 1982, Lyle Johnson, WA7GXD, and Den Connors, KD2S, initiated a packet contact with the first TAPR unit. The project progressed from these first prototype units to the TNC-1 and then finally to the TNC-2 which is now the basis for most amateur packet operations worldwide.

TAPR was founded in 1982 as a national organization with interests in the areas of packet and digital communications. Today, TAPR continues as a membership supported non-profit amateur research and development organization. TAPR currently has more than 2000 members, worldwide. TAPR continues to develop kits for the amateur community and is working actively on publications and communications standards.

Membership:

Membership in TAPR includes a subscription to the quarterly published

Packet Status Register newsletter. PSR has been published since 1982 and is recognized as an authoritative source for up-to-date user and technical information on packet radio. Much of the material in PSR is timeless. Back issues may be obtained from the TAPR office. Membership in TAPR is \$15 for one year. In Canada and Mexico membership is \$18 and outside North America they are \$25.

Join the amateur digital-revolution -- Join TAPR!

Special Interest Groups

TAPR maintains several SIGs in various areas. To get the latest list of SIG groups, send an e-mail message to listserv@tapr.org with the subject line "Request" and the following text in the body of the message:

list (for list of mail groups on TAPR.ORG)

To request the information on any mail group, send an e-mail message to listserv@tapr.org and in the message text include:

information listname (where listname is the name of the mail group)

When you are ready to subscribe, send e-mail to listserv@tapr.org with the following command in the message text:

subscribe <list> <your name>

List should be the name of the mail list you want to join and Your Name should be both your First and Last Name.

Current SIGs include:

NETSIG	Regional Networking SIG
BBSSIG	BBS SIG
HFSIG	SIG on HF Digital Topics
DSP	DSP Software Development Forum
DSP-93	SIG for those that are building TAPR/AMSAT DSP-93 kits
APRSSIG	SIG on APRS

FTP users

The TAPR Software Library is now available via anonymous FTP. You can access the library by ftp access to 'ftp.hereford.ampr.org' in the directory pub/hamradio/tapr. Login in as 'anonymous', with a password of 'your_account@internet_address'.

Packet Raido and TAPR is on the World-Wide-Web!

The series is accesible through the Packet Radio Home Page at this URL:

<http://bb.iu.net/infomotion/taprhomes.html>

<http://bb.iu.net/infomotion/pkthomes.html>

What's here? A page of links (the Home Page), the TAPR Home Page, a page of Packet Pioneers and Contacts, a hypertext packet primer and FAQ, and more.

Thanks to Howie, N2WX, President of Infomotion, for his support and invaluable contributions of the TAPR page.

Tucson Amateur Packet Radio
8987-309 E. Tanque Verde Road #337
Tucson, AZ 85749-9399
817-383-0000 (Tuesday - Friday 9:00-11:00am, 3:00-5:00pm CT)
(15:00-17:00, 21:00-23:00 UTC)
817-566-2544 FAX

Internet: TAPR@TAPR.ORG

6. System Design In Africa

From: Eric Rosenberg <ericr@lan.vita.org>
Date: Wed, 4 Jan 1995 16:49:57 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.3.89.9501041620.B14644-0100000@lan.vita.org>
Rendered body type: text/plain

VITA, Volunteers in Technical Assistance, a non-profit development organization based in Arlington, VA is looking for a qualified volunteer consultant to design a terrestrial digital radio network in Guinea, west Africa.

VITA presently operates nine offices in Guinea, most of which are already connected by a voice (radio) network.

The consultant will meet with the project staff at the Country Office in Conakry to determine their needs and requirements, and then travel to some (or all) of the field offices to perform a first-hand site survey.

Upon returning home, the consultant will provide VITA with a written report outlining the system needs with respect to hardware and software, detail the network architecture, and make recommendations on equipment and software to be installed at a future date. This is **not** an amateur radio network. Familiarity with amateur radio among the project staff in Guinea is, at best, slight.

The trip will last two weeks, and will begin in late January or early February. VITA will be responsible for all costs and expenses related to this trip, but is not in a position to pay a professional fee.

The consultant must have experience in the following areas: HF voice and digital communications; system and network design; familiarity with existing hardware and software; working knowledge of French;

If you are interested or require additional information, please contact Eric Rosenberg at the e-mail address below.

Eric Rosenberg
Volunteers In Technical Assistance voice: +703-276-1800
1600 Wilson Blvd., Suite 500 fax: +703-243-1865
Arlington, VA 22209 USA ericr@vita.org

7. Re: [HFSIG:176] Re: subscribe

From: Greg Jones <gjones@tenet.edu>
Date: Wed, 4 Jan 1995 18:51:31 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199501050051.SAA00858@Leslie-Francis.tenet.edu>
Rendered body type: text/plain

Send email to listserv@tapr.org

in the message, say

get tapr taprinfo.txt

Cheers - Greg

According to Stephen Hall:

>
> As I've forgotten, I too would like this posted. I have some friend that
> would like to subscribe additionally.
>
>
>
> On Wed, 4 Jan 1995, Dan Miller wrote:
>
> > Could someone please email me how to subscribe to this list?
> > Dan dlmill@mtu.edu
> >
> >
> >
> >
>

8. Re: [HFSIG:163] Re: Ideas for HF modulation and coding

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 4 Jan 1995 18:29:09 -0800

To: FORRERJ@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199501050229.SAA11090@servo.qualcomm.com>
Rendered body type: text/plain

>1) (ODFM) n-parallel carriers with orthoganal signaling. All carriers
>are always present. Demodulation is performed by FFT's. It is obvious that
>the choice of the FFT size and sample rate have to just right otherwise we
>will end up with an odd distribution of our carriers over the
>frequency/phase bins. After this choice have been made, how do I use this
>I/Q recovered data? I have some wild ideas, but would like to obtain
>further input first.

Nothing says you have to demod this with an FFT, although that's one way to do it. Once you determine carrier phase and symbol timing, you just generate a bunch of local carrier references, multiply each by the received signals, and integrate and dump or however you want to do the matched filtering. Of course, if you pick the right numbers you would make it easier to do this with an FFT.

>2) (Phil's method based on Welch functions) n-parallel tones with orthoganal
>signaling. Only a small subset of carriers are present at a time - the
>different subset is selected from a Hamadard matrix. In a sense,

Actually, except for one codeword of all zeros the orthogonal functions all have 50% of their bits on and 50% off.

>Piccolo may be considered a special case of this idea, where only one
>carrier is sent at a time from a set of m-carriers. Phil mentioned that FHT
>(is that for fast hadamard transform?) is used to perform demodulation. How
>about telling us a bit more about how to implement this one Phil.

As I understand it, piccolo only sends one tone at a time. What I've proposed is the same thing except for an additional layer of orthogonal signalling in the form of the walsh functions on top of the orthogonal tone set. This is the layer you'd decode with the FHT.

Whether or not this extra layer helps in a real HF channel is unknown.

Phil

9. Re: [HFSIG:165] Re: HF Modulation Modes

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 4 Jan 1995 20:57:48 -0800
To: k5yfw@sacdm10.kelly.af.mil
Cc: hfsig@tapr.org
Message-ID: <199501050457.UAA11138@servo.qualcomm.com>
Rendered body type: text/plain

> Do we want to design a modem for the worse-case scenario or
> can we live with one that can get data thru on the worst
> channel part of the time. I have a pick-up without 4 wheel
> drive...there are at times places I can't drive...so I don't
> drive there until conditions are better. There are places
> where I can drive without 4 wheel drive that my Geo Metro can
> never go. Should I have 4 wheel drive? Do we need a modem
> that can get thru even the worst conditions or can we settle
> for something less. Are we purist or realists?

The newer telephone modems are highly adaptive to line quality. Perhaps *too* adaptive. Lots of different baud rates, symbol constellations and coding rates. It helps, but it sure does complicate things if you're not careful.

Phil

10. Re: [HFSIG:169] Re: HF Modulation Modes

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 5 Jan 1995 02:49:40 -0800
To: jmorriso@bogomips.ee.ubc.c
Cc: hfsig@tapr.org
Message-ID: <199501051049.CAA08743@unix.ka9q.ampr.org>
Rendered body type: text/plain

>This isn't related to HF, but: would some modulation like Piccolo (I
>think I missed the description for that) be applicable to sending a
>data stream over a 2m or 70cm narrow-band FM voice radio? And
>preferably through a voice repeater as well. Speed would be about 50
>bps or so, just enough to send your callsign or a very short message
>when you key up the radio.

Not really. The key to your question is the phrase "narrow-band". Piccolo, and other m-ary orthogonal schemes, are specifically designed to spend bandwidth in order to save on required S/N ratio.

Automatic transmitter IDing is already quite easily done with conventional binary modems. I think MSK is fairly popular for this purpose on public service radios. At least you hear it all the time on cop shows.

Phil

11. Re: HF modulation

From: Eric E Silbaugh <esilbaug@afit.af.mil>
Date: Sat, 7 Jan 1995 13:10:19 -0500
To: hfsig@tapr.org
Cc: esilbaug@afit.af.mil
Message-ID: <199501071810.AA08932@stealth.afit.af.mil>
Rendered body type: text/plain

I have been following the discussion on OFDM and studying Phil's intriguing proposal for using M-ary orthogonal FSK with Walsh functions. Over Christmas break (I'm pursuing an MSEE; if only I could catch it) I did some crude simulations of Phil's scheme using MATLAB.

First, I generated a 64x64 Hadamard matrix (whose rows are Walsh functions). Then I generated 64 equal amplitude sinusoids, each separated by 32 Hz, from 512 Hz to 2528 Hz. I then picked a row at random from the Hadamard matrix to determine which sinusoids to sum together. I sampled the time sequence at 32 kHz and plotted it over a full period of the combined signal.

I used several different Walsh functions to get a feel for the general behavior of these signals. Since I started with 64 tones, and only 1/2 are on for any particular Walsh function, I was combining 32 separate tones. The results were rather interesting, and very similar to Johan's.

Thus wrote Johan, KC7WW:

> As a way to learn more about how OFDM works for the
> 39-tone MIL-STD-188 parallel modem, I wrote a C program
> that simulates 39 parallel tones and combines them into
> a single output signal.

[snip]

> I also did capture the simulated output waveform and was
> quite concerned after inspecting the plot - the
> presence of large spikes of significant magnitude (in
> some cases, some 10-15 dB over the RMS value).

My simulation produced signals with peak-to-RMS ratios of 9 dB, they were generally of low amplitude except where the sinusoids came into phase and produced a large peak, and they generally looked 'noiselike'. I also noticed that the signal's peak amplitude was not symmetric about the time axis. The signals had zero average value but the largest positive-going peak was twice as high as the largest negative-going peak.

As several others have noted, this high peak-to-RMS ratio will limit the amount of energy per symbol. A signal such as this is potential trouble in any amplifier unless care is taken to control the peak level. The asymmetry in peak heights may further limit the amount of power an amplifier could deliver. This is definitely not a constant envelope signal so linear amplifiers are in order. As more tones are added the signals will have larger peak-to-RMS ratios. This is consistent with Johan's observations since he was summing more tones.

The FCC limits amateurs based on PEAK envelope power (PEP), not RMS power. As Phil noted, coding gains can buy back most of the loss in energy per symbol caused by the low RMS value of the signal, but it's still a significant limitation. Disregarding interference, it's ultimately the energy per bit that determines the performance of a digital system.

Phil's proposal was for a noncoherent system, which got me thinking (scary, eh?). A noncoherent system ignores the phase of a signal; so, we have a free parameter to play with. My initial simulation started each sinusoid at zero phase. If we could use the phase to reduce the peak value of the signal we could get more energy per symbol and get the coding gain as well.

First I tried a linear shift in the initial phase across all 64 sinusoids. I started with zero initial phase at the lowest frequency and increased the phase linearly to two pi at the highest frequency. This resulted in signals with about the same peak-to-RMS ratios (8.25 dB), still 'noiselike', but the amplitude peaks became symmetrical about the time axis (interesting and unexpected).

My next attempt was to assign a random initial phase to each of the 64 sinusoids. The initial phase was distributed uniformly between zero and two pi. This produced signals with an average peak-to-RMS ratio of 4.75 dB, the signals looked extremely 'noiselike', and the amplitude peaks were nearly symmetrical. Progress!

These were just the first ways I thought of to change the phase, there was no method to my madness. Even a naive method like randomizing the phase produced a 4 dB reduction in the peak-to-RMS ratio. These results lead to a conjecture and several questions.

My conjecture is that we can reduce the peak-to-RMS ratio further, and maybe even approximate a constant envelope signal, by properly choosing the phases of the sinusoids. I think this is plausible since a wideband FM signal is a constant envelope signal and its spectrum looks like a set of nearly equal amplitude sinusoids, within a limited bandwidth. I doubt we could get a truly constant envelope signal because we are using a small set of harmonically related sinusoids.

Some questions I have:

- Are there systematic methods for choosing the initial phases of the sinusoids to minimize the peak-to-RMS ratio?
- What limits are there on reducing the peak-to-RMS ratio?
- Are these limits, if they exist, related to the number of sinusoids or the Walsh function used?
- Is this a well-known (or obscure) problem that already has a solution?
- Is reducing the peak-to-RMS ratio really as important as I seem to think it is?

The MIL-STD modems apparently use PSK on the tones as well. Since the phase of any particular sinusoid in my simulation was constant over the symbol period this should allow using methods such as DPSK on the tones if someone wants to add an additional layer of modulation. Would this increase the peak-to-RMS ratio?

I can provide the MATLAB M-files I used for this simulation if anyone is interested in duplicating, or extending, my results. Thoughts anyone?

Eric, N2NNP

```

      .:..: .:..: .:..:
      : ' : '
      :..: :..: '
      :: :: '
      : : :
      '::::::::::'

```

Eric E. Silbaugh
esilbaug@afit.af.mil

All standard, non-standard, and
MIL-STD disclaimers apply

12. Re: [HFSIG:183]

From: Larry Gadallah <larry@owrlakh.unbc.edu>
Date: Sat, 7 Jan 1995 14:04:56 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199501072204.OAA02948@owrlakh.unbc.edu>
Rendered body type: text/plain

Eric E Silbaugh <esilbaug@afit.af.mil> writes:

>
> I have been following the discussion on OFDM and studying
> Phil's intriguing proposal for using M-ary orthogonal FSK
> with Walsh functions. Over Christmas break (I'm pursuing
> an MSEE; if only I could catch it) I did some crude
> simulations of Phil's scheme using MATLAB.
>
> First, I generated a 64x64 Hadamard matrix (whose rows are
> Walsh functions). Then I generated 64 equal amplitude
> sinusoids, each separated by 32 Hz, from 512 Hz to 2528 Hz.
> I then picked a row at random from the Hadamard matrix to
> determine which sinusoids to sum together. I sampled the
> time sequence at 32 kHz and plotted it over a full period
> of the combined signal.
>

Can anyone provide a good reference to what some of these things are (i.e. Walsh functions, Hadamard matrices)?

Thanks and 73,

--

Larry Gadallah	Prince George, BC
Internet: larry@owrlakh.unbc.edu	TPC: (604) 964-1140
VE4TCP/VE6VQ	ICBM: 53:51'38.4"N 122:44'25.8"W

13. G-TOR Goes Public

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Mon, 9 Jan 95 08:26:43 -0600
To: hfsig@tapr.org
Message-ID: <m0rRL3U-0002CeC@sacdm10.kelly.af.mil>
Rendered body type: text/plain

G-TOR Goes Public

by Walt DuBose, K5YFW

While at the local "ham store" on Christmas Eve, my XYL WB5WXY, suggested that I buy a "ham book" (she meant magazine I'm sure) to read while we were flying to Idaho Christmas day.

While looking at the various "books", I spotted "G-TOR, The New Mode". My wife was a little surprised at the \$16 price but I assured here this would keep me occupied and it has.

The cover says "Articles, Charts, Protocol". Copyright by Kantronics, this 90 page book is crammed with articles and papers on G-TOR. While many articles are redundant, reading several articles will give you a good working knowledge of G-TOR and why Kantronics is proud of their product.

Not only is the book filled with articles, it also contains charts and a complete description of the protocol.

I will have to re-read several articles to ensure that I fully understand this protocol. For those with programming background, I feel confident that they will get much more out of this book than a non-technical person like me.

While not everyone will want to rush out and purchase this book, those truly interested in high speed HF data communications will want a copy. This is also an excellent reference for your radio club's library.

14. Re: G-TOR Booklet

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Mon, 9 Jan 95 10:20:49 -0600
To: rs@ife.ee.ethz.ch
Cc: hfsig@tapr.org
Message-ID: <m0rRMpu-0001zfC@sacdm10.kelly.af.mil>

Rendered body type: text/plain

I'm sorry, I guess I should have included the information. The title of the book is "G-TOR, The New Mode", copyright and published by:

Kantronics Company, Inc
1202 E. 23rd St.
Lawrence, KS 66046

U.S. Telephone Number: 913/842-7745

I believe you can only get it from Kantronics or through one of their equipment distributors.

73,

Walt

In your message of 9 Jan 1995 at 1613 CST, you write:

> Hi Walt

>

> Thanks for your very interesting hint.

> Could you please give us (overseas) more info about publisher/publishing date and place
> and so on, so that we can order it by a local book-store.

> Maybe this booklet even has an international book number (in the form of ISSN-...) or
> maybe there is an indication/address/fax-number where we could purchase it ?

>

> (I am probably not going to implement that protocol on my DSP (as I did with PACTOR), but
> would also enjoy the reading about the algorithms behind).

>

> Thanks for your help es vy 73s,

> Rolf

> --

>

--- * * * ---

>

> Rolf Sommerhalder

> Swiss Federal Institute of Technology (ETH)

> Electronics Laboratory ETZ H60.1

> Gloriastrasse 35

> 8092 Zurich / Switzerland

>

> Voice +41 1 632 76 40 (or 632 51 53)

> Fax +41 1 632 12 10

> E-Mail rolf.sommerhalder@ife.ee.ethz.ch

>

home address:

c/o Rindlisbacher

Wermatswilerstrasse 96

8610 Uster / Switzerland

Voice +41 1 941 59 28

AX.25 HB9CWP @ HB9OS

Swiss Disaster Relief HBK81

15. Re: [HFSIG:186] Re: G-TOR Booklet

From: Stephen Hall <shall@rain.org>

Date: Mon, 9 Jan 1995 08:59:11 -0800 (PST)

To: hfsig@tapr.org

Cc: Ross duClair <CPTRoss@aol.com>

Message-ID: <Pine.SUN.3.90.950109085422.1092A-100000@coyote.rain.org>

Rendered body type: text/plain

Rolf, Regarding G-TOR: I attended a talk given by Phil Anderson regarding G-TOR vs. Pactor and it appeared that G-TOR has much more powerful error correction than Pactor. I would be interested in your evaluation after you have had an opportunity to check it out. Steve, WM6P

On Mon, 9 Jan 1995, WALT DUBOSE - K5YFW wrote:

> I'm sorry, I guess I should have included the information. The title
> of the book is "G-TOR, The New Mode", copyright and published by:

>

> Kantronics Company, Inc

> 1202 E. 23rd St.

> Lawrence, KS 66046

>

> U.S. Telephone Number: 913/842-7745

>

> I believe you can only get it from Kantronics or through one of their
> equipment distributors.

>

> 73,

>

> Walt

>


```

>
> In your message of 9 Jan 1995 at 1613 CST, you write:
> > Hi Walt
> >
> > Thanks for your very interesting hint.
> > Could you please give us (overseas) more info about publisher/publishing date and place
> > and so on, so that we can order it by a local book-store.
> > Maybe this booklet even has an international book number (in the form of ISSN-...) or
> > maybe there is an indication/address/fax-number where we could purchase it ?
> >
> > (I am probably not going to implement that protocol on my DSP (as I did with PACTOR), but
> > would also enjoy the reading about the algorithms behind).
> >
> > Thanks for your help es vy 73s,
> > Rolf
> > --
> >
> >                                     --- * * * ---
> >
> > Rolf Sommerhalder
> > Swiss Federal Institute of Technology (ETH)      home address:
> > Electronics Laboratory ETZ H60.1                c/o Rindlisbacher
> > Gloriastrasse 35                               Wermatswilerstrasse 96
> > 8092 Zurich / Switzerland                      8610 Uster / Switzerland
> >
> > Voice +41 1 632 76 40 (or 632 51 53)           Voice +41 1 941 59 28
> > Fax   +41 1 632 12 10                         AX.25 HB9CWP @ HB90S
> > E-Mail rolf.sommerhalder@ife.ee.ethz.ch         Swiss Disaster Relief HBK81
> >
>

```

16. Re: HF modulation

```

From: BLOOM_ALAN/HP5300_R0@opnmail3.corp.hp.com
Date: Mon, 9 Jan 95 11:11:00 -0800
To: hfsig@tapr.org
Message-ID: <d0ey1Sy00000000000@MHS>
Rendered body type: text/plain

```

Eric Silbaugh wrote:

(Re: OFDM)

>My simulation produced signals with peak-to-RMS ratios of
>9 dB, ...

>As several others have noted, this high peak-to-RMS ratio
>will limit the amount of energy per symbol. ...

>The FCC limits amateurs based on PEAK envelope power (PEP),
>not RMS power. As Phil noted, coding gains can buy back
>most of the loss in energy per symbol caused by the low RMS
>value of the signal, but it's still a significant limitation.
>Disregarding interference, it's ultimately the energy per
>bit that determines the performance of a digital system. ...

Of course, lower average power also means less interference to
other users. Since the signal will be spread over several kHz,
the energy per 300 Hz channel will be low. Interference *from*
other users (analog or digital) can be reduced with proper coding.

>A noncoherent system ignores the
>phase of a signal; so, we have a free parameter to play with.
>My initial simulation started each sinusoid at zero phase.
>If we could use the phase to reduce the peak value of the
>signal we could get more energy per symbol and get the coding
>gain as well. ...

>My next attempt was to assign a random initial phase to
>each of the 64 sinusoids. The initial phase was distributed
>uniformly between zero and two pi. This produced signals with
>an average peak-to-RMS ratio of 4.75 dB, the signals looked
>extremely 'noiselike', and the amplitude peaks were nearly
>symmetrical. Progress!

>These were just the first ways I thought of to change the
>phase, there was no method to my madness. Even a naive
>method like randomizing the phase produced a 4 dB reduction
>in the peak-to-RMS ratio. ...

The "naive method" may be pretty close to optimal. Since the

number of cycles per symbol of each carrier differs from each of the others by an integer number of cycles, each pair of carriers will be in phase (3 dB peak-to-average) sometime during each symbol. Intuitively, it seems like you could never do that well with many carriers.

Al Bloom N1AL

17. Re: [HFSIG:188] Re: HF modulation

From: "Johan Forrer"
Date: Mon, 9 Jan 1995 11:57:49 PST8PDT
To: hfsig@tapr.org
Message-ID: <D33ED2758A7@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Eric E. Silbaugh wrote:

>I have been following the discussion on OFDM and studying
>Phil's intriguing proposal for using M-ary orthogonal FSK
>with Walsh functions. Over Christmas break (I'm pursuing
>an MSEE; if only I could catch it) I did some crude
>simulations of Phil's scheme using MATLAB.

>First, I generated a 64x64 Hadamard matrix (whose rows are
>Walsh functions). Then I generated 64 equal amplitude
>sinusoids, each separated by 32 Hz, from 512 Hz to 2528 Hz.
>I then picked a row at random from the Hadamard matrix to
>determine which sinusoids to sum together. I sampled the
>time sequence at 32 kHz and plotted it over a full period
>of the combined signal.

.....snip.....

> - Are there systematic methods for choosing the
> initial phases of the sinusoids to minimize the
> peak-to-RMS ratio?

Eric, just a suggestion: you may want to look at an earlier (way back) message thread on this very topic. It is recorded in an archive on ftp.ucsd in the DSP directory. If you have a problem obtaining it, I'll dig out the name of the archive.

I think there is no good news here. There is no easy solution as far as I can gather (?). CLOVER, as an example, addressed this problem by using time and frequency interleaved signaling as well as pulse shaping.

..... more snipping

>The MIL-STD modems apparently use PSK on the tones as well.
>Since the phase of any particular sinusoid in my simulation
>was constant over the symbol period this should allow using
>methods such as DPSK on the tones if someone wants to add an
>additional layer of modulation. Would this increase the
>peak-to-RMS ratio?

I suspect it depends on the peak of the individual channel waveform.

>I can provide the MATLAB M-files I used for this simulation
>if anyone is interested in duplicating, or extending, my
>results. Thoughts anyone?

I, for one, am quite interested in your work, Eric. Likewise, if anyone is interested in the OFDM simulation, I would be happy to share the "C" source code.

A bit of progress over the last weekend: I finished a FFT-based display scope program for the DSP sound card. Since I am interested in using the FFT-approach for doing an OFDM modem, thought that I could use the same FFT for different purposes. It's a 256-point radix-4 that includes a Hanning window (and log transform for calculations in dB, if needed) and does it in approximately 250 micro seconds. The first thing I did was to look at the audio frequency responses of my ICOM 751 - boy! was I surprised.

I went ahead and played with the DSP sound card and implemented the parallel tone modulator and FFT-based OFDM demodulator - it is surprising to see how sensitive the FFT is to any clipping effects of these large spikes in the signal. One has to be real careful of this, else a bit of garbage seem to appear in the wrong frequency bins. However, when this kind of garbage ends up in the wrong frequency bin, it seems to appear as uncorrelated noise - the actual signal in the frequency bin still appears quite distinct and recoverable. Perhaps this side-effect is not all that serious after all. Comments?

73's

Johan, KC7WW

18. HF DSPs

From: "Victorio O. Ochave" <jojo@asti.dost.gov.ph>
Date: Tue, 10 Jan 1995 17:59:20 +0800
To: hfsig@tapr.org
Cc: jojo@christine.asti.dost.gov.ph
Message-ID: <199501100959.RAA04238@christine.asti.dost.gov.ph>
Rendered body type: text/plain

I'm a new member of the list, and also new in data comm over HF. We will be establishing some HF links here in the Philippines using GTOR. I was told that there are DSP filter units (Timewave DSP 59+) useful for PACTOR and AMTOR links. I was wondering if the same DSP units can make a substantial improvement in performance when using the GTOR protocol.

All commentaries and advise on this will be highly appreciated.

Victorio A. Ochave, Jr.
Communications Engineering Division
Advanced Science and Technology Institute (ASTI)
Department of Science and Technology (DOST)
4/F NEC Bldg., University of the Philippines
Diliman, Quezon City, PHILIPPINES 1101

voice: 632 995071-9 loc.5106
fax: 632 9224714
Internet: jojo@asti.dost.gov.ph
ochave@itu.ch
v.ochave@ieee.org
X.400: G=victorio; S=ochave; P=itu; A=arcom; C=ch

19. Re: [HFSIG-190] HF DSPs

From: "Rick Whiting" <Rick_Whiting@ATK.COM>
Date: 10 Jan 1995 10:39:11 -0600
To: hfsig@tapr.org
Message-ID: <199501101634.KAA18813@ATK.COM>
Rendered body type: text/plain

Reply to: RE:[HFSIG:190] HF DSPs
Date: 1/10/95 5:14 AM
To: Rick Whiting
From: hfsig@tapr.org
Victorio A. Ochave, Jr. <jojo@asti.dost.gov.ph> wrote:

>I'm a new member of the list, and also new in data comm over HF. We >will be establishing some HF links here in the Philippines using GTOR. I >was told that there are DSP filter units (Timewave DSP 59+) useful for >PACTOR and AMTOR links. I was wondering if the same DSP units can >make a substantial improvement in performance when using the GTOR >protocol.

>All commentaries and advise on this will be highly appreciated.

Victorio, If the AF DSP filter helps for PACTOR and AMTOR it will similarly help for G-TOR. However, it is highly desired to develop the selectivity as high up in the receiver's IF as practical, rather than at AF. If the IF filters are too wide relative to the desired signal bandwidth, noise and interference outside the AF passband will still impact the radio, e.g., the AGC, and degrade reception even though you don't "hear" it out of the AF (DSP) filter.

Unfortunately, many folks are trapped into using the too-wide filters that are supplied with their transceivers for HF data modes like PACTOR and G-TOR.

And since the filters were not designed for data with respect to phase linearity, especially at the edges, the stock filters with about the "right" bandwidth may not work well either! Thus, in many practical applications the DSP filter at AF significantly improves performance.

By the way, I think the Timewave DSP filters are top notch.

73/Rick W0TN

20. Re: [HFSIG:173] HF Modulation

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 10 Jan 1995 17:59:18 -0800
To: hfsig@tapr.org
Message-ID: <199501110159.RAA25908@servo.qualcomm.com>
Rendered body type: text/plain

Johan says:

>Bob asked an interesting question: whether soft decision decoding
>with convolutional coding was as effective as linear block coding
>with interleaving.

I'd been meaning to comment on this, glad you reminded me.

I've learned that this has been a rather controversial subject that some people argue with a fervor that approaches the classic VC vs datagram wars of 10 years ago.

Up front I should say that I happen to work for one of the main parties to this debate (Andrew Viterbi) so my comments may be biased.

Convolutional coding has the following general advantages:

1. Better performance for equivalent code complexity
2. Can use "soft decision" information for improved performance (2dB on AWGN) much more easily
3. More easily adapted to arbitrary block sizes

Block codes, as typified by Reed Solomon codes, have the following general advantages:

1. No performance penalty for "systematic" codes (where the data appears unchanged as a subset of the coded output)
2. Complex codes more easily implemented (partially negating convolutional advantage #1)
3. No need to "tail off" blocks for full coding gain
4. Less sensitive to burst errors
5. Direct adaptability to higher-order modulation symbol alphabets, such as the m-ary FSK schemes I've been discussing.

Nobody really disputes the correctness of any of these points, since they're amenable to mathematical proof. The arguments all center over the relevance of their assumptions, and which advantages are more important in practice. For example, both Viterbi and Reed-Solomon decoders are amenable to VLSI implementation, but differences in the details can give one a temporary advantage over the other.

It's also common to combine convolutional and block codes to get the best of both worlds. These "concatenated" coding systems are commonly used in deep space communications, starting with Voyager. You run a Reed-Solomon block code on top of a Viterbi-decoded convolutional code, usually with an interleaver between the two.

The Viterbi decoder does most of the work, and since it's directly connected to the modem it can benefit from soft decision samples. But when the Viterbi decoder is temporarily overwhelmed, it emits bursts of "hard" (binary) errors that are well adapted to the capabilities of the Reed Solomon decoder.

The Voyager concatenated coding system is fairly simple by modern standards and is well within our capability. I've already done the Viterbi decoder for the NASA standard $r=1/2$ $K=7$ convolutional code.

N9FZX has done a Reed-Solomon decoder that may be the same one that Voyager uses, but I'm not sure. In any event, all you need to combine the two is some glue (interleaving, buffering, etc). The waterfall curve for the Voyager coding scheme is a nearly vertical wall at around 2-2.5dB.

Galileo uses the same basic techniques, but taken to some wild extremes to squeeze the most out of the omnidirectional antenna since the failure of the main antenna. The convolutional code has $r=1/4$, $K=14$, which is extremely high for a Viterbi decoder (the work factor increases exponentially with K). And the RS code uses varying amounts of redundancy per block which permits a trick called "redecoding".

First you run the Viterbi decoder. Then you run the RS decoder on the output. Some of the RS blocks will decode, some won't; the blocks with more redundancy being more likely to decode. You fill in the missing symbols in the decoded blocks and then toss them back to the Viterbi decoder for another go with the RS-decoded states "pinned down". Then you have a second go at the RS decoder. You iterate this perhaps 4-5 times until things stop improving. The end result is an astonishingly low E_b/N_0 requirement of about 0.56dB, damn near the Shannon limit for that channel. Of course, they do this at only 10 b/s on a Sparcstation...

Phil

21. Re: [HFSIG:192] Re: HF Modulation

From: "Johan Forrer"
Date: Wed, 11 Jan 1995 08:50:14 PST8PDT
To: hfsig@tapr.org
Message-ID: <D60CDC50F6D@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Phil,

That essay on coding was welcomed by everyone - well done!

I would just like to add, if I may, one further thought relating specifically to the multitone signaling systems and coding methods. In a previous posting I mentioned the work by Pierce et, al. - nobody seems to have commented on that. Before it is lost - here is a brief summary:

The fact that was established in this research paper, that was based on actual transatlantic and transcontinental HF data links at various HF frequencies, was that HF disturbances has somewhat characteristic nature, either broadband and/or noise bursts in conjunction with various forms of fading. This often has some frequency/sequency effects that could be put to good use in the type of coding applied, i.e. the choice of interleaving/constraint length for example.

The second fact that the research established, was that channel spacing, (i.e. signaling over orthogonally spaced channels) appears to have a systematic effect on BER. For example, if you look at BER vs. coding scheme, cyclic behavior with distinct minima is observed - that tells us that it is possible to optimise coding gain by proper choice of parameters, such as constraint length, interleave factor, and type of code.

Taking these observations into account, as I have indicated in my earlier posting - an appropriate convolutional coder or the right type of block code was equally effective on parallel tone modems. This reseach was done prior to soft-decision coding becoming popular and I am sure other advances were made in coding theory since.

There is no question that a combined block/convolutional code, possibly with soft-decision coding, will buy us the best of both worlds. Coding theory, however, is not one of the easiest of fields to wrestle with - for one, it does not allow the same degree of intuition or gut feeling to be applied as for instance modulation/demodulation theories. You really have to know what you are doing.

I think we all agree that we are very much dependant on the guidance and contributions of experts like Phil.

Looking forward to learning more about this field.

73's

Johan, KC7WW

22. voice?

From: jmorriso@bogomips.ee.ubc.ca (John Paul Morrison)
Date: Wed, 11 Jan 1995 11:03:15 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0rS8KB-0004gbC@bogomips.ee.ubc.ca>
Rendered body type: text/plain

Will these 2400bps HF modems be able to work with digitized+compressed voice? (CELP or something). That would really be something to impress the regular (digital is for weenies) HFers! The answer should be 'yes' if the BER is reasonable and it would be 'usable' if there's no long training sequence between modems (I mean you couldn't key up and say 'QSL' or something short if it took 10 seconds to train every time).

I don't know what's available in hardware voice compression etc., but the ICs are no doubt getting cheap.

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
jmorriso@bogomips.ee.ubc.ca ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org

23. Re: [HFSIG:190] HF DSPs

From: shall@rain.org (Steve Hall)
Date: Wed, 11 Jan 95 12:36:59 PST
To: hfsig@tapr.org
Message-ID: <9501112036.AB19335@coyote.rain.org>
Rendered body type: text/plain

Victorio,

If you would care to run any on the air tests on G-TOR to Southern California please let me know.

Steve Hall, WM6P

>I'm a new member of the list, and also new in data comm over HF. We will be establishing some HF links here in the Philippines using GTOR. I was told that there are DSP filter units (Timewave DSP 59+) useful for PACTOR and AMTOR links. I was wondering if the same DSP units can make a substantial improvement in performance when using the GTOR protocol.

>

>All commentaries and advise on this will be highly appreciated.

>

>

>

>

>-----
>Victorio A. Ochave, Jr.

>Communications Engineering Division

>Advanced Science and Technology Institute (ASTI)

>Department of Science and Technology (DOST)

>4/F NEC Bldg., University of the Philippines

>Diliman, Quezon City, PHILIPPINES 1101

>

>voice: 632 995071-9 loc.5106

>fax: 632 9224714

>Internet: jojo@asti.dost.gov.ph

>ochave@itu.ch

>v.ochave@ieee.org

>X.400: G=victorio; S=ochave; P=itu; A=arcom; C=ch

>

>-----

>

>

>

Steve Hall

24. Software tools for soundcard

From: "Johan Forrer"
Date: Wed, 11 Jan 1995 12:26:08 PST8PDT
To: HFSIG@tapr.org
Message-ID: <D6467482522@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

For those folks working or planning to work with the PSA sound cards, here

is some great news.

According to a post on Usenet, version 5.01 of the Analog Devices' software development tools for the 21xx DSP is available via anonymous ftp. This toolkit is professional quality and include, amongst other things, the GNU C compiler, GNU assembler, linker, prom splitter, and simulator. Since I have purchased this software some time ago (without knowing that it was on the net), and used it extensively, I can highly recommend it.

Only minor changes are necessary to the software that I have previously made available. In fact, the host code requires only commenting out one line of code. The DSP code requires a few changes - though nothing serious. The fact that you can now concentrate on your DSP application instead of concerning yourself with how and where stuff gets loaded and cross-linked, makes programming the DSP a pleasure.

The software tools are available at: ftp.analog.com in directory /pub/dsp/devtools/21xx. It's some 2.8 M, so beware. I downloaded it yesterday (1/10/95) to make sure it was'nt a hoax - its real OK, but I'm not sure whether this is an accident?

Trust this is of interest.

73's

Johan, KC7WW

25. Re: [HFSIG:194] voice?

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Wed, 11 Jan 95 14:50:14 -0600
To: hfsig@tapr.org
Message-ID: <m0rS9zi-0002BuC@sacdm10.kelly.af.mil>
Rendered body type: text/plain

The MIL-STD-188-110A (39 parallel tone) modems were designed to do just that. In fact, the first 39 parallel tone modem I ever saw (in about 86 or 87) was in a voice encryption device. I believe the military used LCP10 voice encryption. -- k5yfw

In your message of 11 Jan 1995 at 1342 CST, you write:

>
> Will these 2400bps HF modems be able to work with digitized+compressed
> voice? (CELP or something). That would really be something to impress
> the regular (digital is for weenies) HFers! The answer should be 'yes'
> if the BER is reasonable and it would be 'usable' if there's no long
> training sequence between modems (I mean you couldn't key up and say
> 'QSL' or something short if it took 10 seconds to train every time).
>
> I don't know what's available in hardware voice compression etc., but
> the ICs are no doubt getting cheap.
>
>
>
> -----
> BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
> jmorriso@bogomips.ee.ubc.ca ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org
> -----
>

26. (no subject)

From: (unknown sender)
Date: (date unavailable)
Rendered body type: text/plain
(Smail3.1.29.1 #5) id m0rSBG4-0001L5C; Wed, 11 Jan 95 16:11 CST
Received: from grouse.afit.af.mil by stealth.afit.af.mil with SMTP id AA24672
(5.65c/IDA-1.4.4 for <hfsig@tapr.org>); Wed, 11 Jan 1995 17:11:09 -0500
Date: Wed, 11 Jan 1995 17:11:09 -0500
From: Eric E Silbaugh <esilbaug@afit.af.mil>
Message-Id: <199501112211.AA24672@stealth.afit.af.mil>
To: hfsig@tapr.org
Subject: Peak Reduction (was RE: HF Modulation)
Cc: esilbaug@afit.af.mil

Al,

I agree, "...lower average power means less interference to other users. ..."

Hi Eric,

What surprised me was to see the different filter responses for "RTTY" and "LSB". RTTY mode gave a fairly steep skirt starting around 1500 Hz, then tapered off, nearly in a linear fashion, down to 4kHz - no "flat top" at all. LSB, on the other hand, contained a band between approximately 300 Hz and 2600 Hz, though not a very clean filter response shape. If, for instance, the full 3kHz is required for a parallel modem, there would be some impact on the channels closer to the band edges.

What I am seeing, of course, is a result of the rig's audio output response at the headphone jack as well as the sound card's audio input response. Perhaps a better place to tap this audio would be at a point before the audio stages.

The tests were mainly to play with the real-time FFT scope and is actually the same DSP code used in the parallel modem demodulator. Fun stuff.

Johan, KC7WW

29. Re: [HFSIG:200] Re: ICOM 751 Filter response curves

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Wed, 11 Jan 95 21:05:25 CST
To: hfsig@tapr.org
Message-ID: <2471@k5yfw.ampr.org>
Rendered body type: text/plain

In message <D67CC315CCE@frl.orst.edu> Johan writes:

> Hi Eric,
>
> What surprised me was to see the different filter responses for "RTTY" and
> "LSB". RTTY mode gave a fairly steep skirt starting around 1500 Hz, then
> tapered off, nearly in a linear fashion, down to 4kHz - no "flat top" at
> all. LSB, on the other hand, contained a band between approximately 300
> Hz and 2600 Hz, though not a very clean filter response shape. If, for
> instance, the full 3kHz is required for a parallel modem, there would be
> some impact on the channels closer to the band edges.

The AN/URC-119 (Harris RF-350) HF military/commercial transceiver has 3 KHz SSB filters...one of each sideband. I have run IF bandpass checks on them and they have a very clean response and quite a sharp cutoff above 3 KHz...much cleaner than amateur equipment as noted above. This might mean that some BW less than 3 KHz would be needed of compensation for the upper 1 KHz of the IF bandpass.

>
> What I am seeing, of course, is a result of the rig's audio output
> response at the headphone jack as well as the sound card's audio input
> response. Perhaps a better place to tap this audio would be at a
> point before the audio stages.

The AN/URC-119 (Harris RF-350) has an audio jack that bypasses the speaker audio stages and mic. amp. stages. The audio there is much cleaner than at the mic. or headphone jacks. This is where they connect the MIL-STD-188-110A (39 tone) modem.

On my IC-735, I have found that the audio jacks on the back panel provide much cleaner audio for data work. I don't know if the IC-751 has such a jack.

>
> The tests were mainly to play with the real-time FFT scope and is actually
> the same DSP code used in the parallel modem demodulator. Fun stuff.
>

The MIL-STD modems require 10 Hz dial resolution read-out but the frequency *MUST* be within about 3 or 4 Hz to the modems to work correctly. That would require some kind of "on-screen" tuning for use with an amateur radio...except for something like the SGC-2000 or Kenwood TKM-707 which are commercial radios with the SGC meeting MIL-Specs.

Walt@home.2.nite

30. More on Galileo coding

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 11 Jan 1995 23:23:22 -0800

To: hfsig@tapr.org
Message-ID: <199501120723.XAA05968@servo.qualcomm.com>
Rendered body type: text/plain

Here's an abstract I found on a NASA web server that summarizes the coding tricks they're using to milk the most out of the low-gain antenna on Galileo. Of course, the deep space channel is very different from the HF channel, but it gives you an idea of what coding can do.

Enhanced decoding for the Galileo S-band mission

DOLINAR, S.; BELONGIE, M.

Jet Propulsion Lab., California Inst. of Tech., Pasadena, CA.

Published: August 1993 Pages: 00016 Biblio Refs: 0

A coding system under consideration for the Galileo S-band low-gain antenna mission is a concatenated system using a variable redundancy Reed-Solomon outer code and a (14,1/4) convolutional inner code. The 8-bit Reed-Solomon symbols are interleaved to depth 8, and the eight 255-symbol codewords in each interleaved block have redundancies 64, 20, 20, 20, 64, 20, 20, and 20, respectively (or equivalently, the codewords have 191, 235, 235, 235, 191, 235, 235, and 235 8-bit information symbols, respectively). This concatenated code is to be decoded by an enhanced decoder that utilizes a maximum likelihood (Viterbi) convolutional decoder; a Reed Solomon decoder capable of processing erasures; an algorithm for declaring erasures in undecoded codewords based on known erroneous symbols in neighboring decodable words; a second Viterbi decoding operation (redecoding) constrained to follow only paths consistent with the known symbols from previously decodable Reed-Solomon codewords; and a second Reed-Solomon decoding operation using the output from the Viterbi redecoder and additional erasure declarations to the extent possible. It is estimated that this code and decoder can achieve a decoded bit error rate of $1 \times 10^{(exp 7)}$ at a concatenated code signal-to-noise ratio of 0.76 dB. By comparison, a threshold of 1.17 dB is required for a baseline coding system consisting of the same (14,1/4) convolutional code, a (255,223) Reed-Solomon code with constant redundancy 32 also interleaved to depth 8, a one-pass Viterbi decoder, and a Reed Solomon decoder incapable of declaring or utilizing erasures. The relative gain of the enhanced system is thus 0.41 dB. It is predicted from analysis based on an assumption of infinite interleaving that the coding gain could be further improved by approximately 0.2 dB if four stages of Viterbi decoding and four levels of Reed-Solomon redundancy are permitted. Confirmation of this effect and specification of the optimum four-level redundancy profile for depth-8 interleaving is currently being done. Author (revised)

NASA Subject Category: COMMUNICATIONS AND RADAR

Major Subject Terms: CONCATENATED CODES, DECODING, GALILEO PROJECT, MAXIMUM LIKELIHOOD ESTIMATES, MICROWAVE ANTENNAS, REED-SOLOMON CODES

Minor Subject Terms: BIT ERROR RATE, CODING, DECODERS, SIGNAL TO NOISE RATIOS, SUPERHIGH FREQUENCIES, SYMBOLS, ULTRAHIGH FREQUENCIES

CASI Accession Number: N941438

Contract Number: RTOP 314-30-61-

Report Number: none

Translation Note: none

Ordering Information: Avail: CASI HC A03/MF A03

Document Class: unknown

31. Re: [HFSIG:174] Re: HF modulation

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 12 Jan 1995 16:42:12 -0800
To: hfsig@tapr.org
Message-ID: <199501130042.QAA15580@servo.qualcomm.com>

Rendered body type: text/plain

> Boy does the signal ever sound like noise...as you tune across
> the signal the S meter will rise and fall and you will think you
> are just tuning a noisy frequency.

A year or so ago somebody on the cypherpunks list came up with the following wonderful adaptation of Clarke's famous saying that "any sufficiently advanced technology is indistinguishable from magic":

"Any sufficiently advanced communication is indistinguishable from noise".

This is actually implied by Shannon's theory. The more efficient a modulation/coding method is, the more noiselike it must become. The reverse is not necessarily true (just because it's noiselike doesn't mean it's efficient).

Phil

32. Re: [HFSIG:193] Re: HF Modulation

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 13 Jan 1995 14:00:25 -0800
To: hfsig@tapr.org
Message-ID: <199501132200.OAA25848@servo.qualcomm.com>
Rendered body type: text/plain

>Taking these observations into account, as I have indicated in my earlier
>posting - an appropriate convolutional coder or the right type of block
>code was equally effective on parallel tone modems. This research was done
>prior to soft-decision coding becoming popular and I am sure other advances
>were made in coding theory since.

Soft-decision decoding wins especially big on fading channels, by the way. As I mentioned in my discussion, convolutional decoding is much more easily adapted to soft decision decoding. It's almost universal in Viterbi decoding, somewhat less so in sequential because of its greater sensitivity to gain errors.

Soft decision decoding can be adapted to block codes, but with considerably greater difficulty. The technique is called "Chasing bits", after David Chase, the inventor. Basically, you determine your N weakest received symbols. Then you try decoding all 2^N possible values of these weak symbols (for a binary code) along with the "strong" received symbols. Take each decoded block, re-encode it and pick the result that's closest in Hamming distance to the original received block of symbols.

Obviously the complexity goes up exponentially with N. That can really stress your block decoding software.

Non-binary codes like Reed Solomon codes can be "softened" somewhat by simply erasing your weakest symbols. Since a RS code can correct twice as many symbol erasures as errors, you stand an improved chance of decoding the block correctly.

Erasure decoding on a binary channel can be seen as soft-decision decoding with only 3 symbol values: "1", "0" and "unknown". A little "softness" goes a long way; just adding an erasure indication is a big win, especially on deep fading channels. In my experiments with rate 1/2 sequential Fano decoding, I found I could barely handle 8-9% symbol errors on a hard-decision binary channel. But if I introduced erasures rather than errors, I could tolerate nearly 50% erasures.

One way to implement erasure decoding even when you don't have a 3-level channel is to send a packet twice and compare the two. You erase any symbols that don't match.

Phil

33. Re: [HFSIG:204] Re: HF Modulation

From: "Johan Forrer
Date: Fri, 13 Jan 1995 20:35:15 PST8PDT
To: hfsig@tapr.org
Message-ID: <D9C90023B27@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Phil,

I have been making good progress with the parallel DSP modem - a few tricks

on the tone generation and it actually now sounds a little musical, perhaps a little warbly - this is an 8 channel version that fits into 250 Hz and has a raw bit rate of 1000 bps - At the moment I'm testing things by running the the modem in full duplex with loopback.

Phil has been making excellent contributions on coding methods - in addition, we also have to start discussing the data link layer in all seriousness. I would suggest that the first attempt be at a simple FEC method but are open to any suggestions?

73's

Johan, KC7WW

34. Re: [HFSIG:205] Re: HF Modulation

From: frode@dxcern.cern.ch (Frode Weierud)
Date: Wed, 18 Jan 1995 16:28:15 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <9501181528.AA25255@dxcern.cern.ch>
Rendered body type: text/plain

Hi all,

> I have been making good progress with the parallel DSP modem - a few tricks
> on the tone generation and it actually now sounds a little musical,
> perhaps a little warbly - this is an 8 channel version that fits into 250 Hz
> and has a raw bit rate of 1000 bps - At the moment I'm testing things by
> running the the modem in full duplex with loopback.

> Phil has been making excellent contributions on coding methods - in
> addition, we also have to start discussing the data link layer in all
> seriousness. I would suggest that the first attempt be at a simple FEC
> method but are open to any suggestions?

It appears to be that fragments of code are growing in several corners of the HFSIG list, such as Johan's channel simulator code, Johan's DSP code for the MIL-STD-188-110A parallel tone modem and Phil's code for the Viterbi algorithm.

I for one would like very much to have a peek at this code to play around a bit, get my hands 'dirty' and get some experience. Would it be possible to arrange an Anonymous FTP server somewhere where the list members could dump their code, data files, documents etc. and where we all can have easy access. I agree that lot of the code will be experimental and not in a form fit for public distribution, but on a UNIX host it is very easy to arrange for a hidden directory to house our not yet consumer ready code.

73 Frode, LA2RL

35. Re: [HFSIG:206] Re: HF Modulation

From: "Johan Forrer
Date: Wed, 18 Jan 1995 08:54:34 PST8PDT
To: hfsig@tapr.org
Message-ID: <E08E528710B@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Frode,

<some lines deleted>

>

>It appears to be that fragments of code are growing in several
>corners of the HFSIG list, such as Johan's channel simulator code,
>Johan's DSP code for the MIL-STD-188-110A parallel tone modem and
>Phil's code for the Viterbi algorithm.

>

>I for one would like very much to have a peek at this code to play
>around a bit, get my hands 'dirty' and get some experience. Would
>it be possible to arrange an Anonymous FTP server somewhere where
>the list members could dump their code, data files, documents etc.
>and where we all can have easy access. I agree that lot of the code
>will be experimental and not in a form fit for public distribution,
>but on a UNIX host it is very easy to arrange for a hidden directory
>to house our not yet consumer ready code.

>

>73 Frode, LA2RL

Thanks for bringing this up - Greg suggested we use tcet.unt.edu in the mean time and I checked and found that there indeed was a TAPR subdirectory. I am not sure what the future plans are for a permanent home for these code fragments.

The bits and pieces that I have is rather experimental and at this stage would like to hang on a bit before placing it on an anonymous ftp site. However, I have been sharing the source code with whoever asked for it, at least I can warn them ahead of time of what they are in for. Please feel free to request the code.

I have a "C" version of the HF channel simulator as well as the beginnings of a DSP version for the sound card DSP. The "C" simulation of the OFDM modem is probably the easiest to understand and allows you to modify it to your liking. The DSP version is more involved as it has to deal with optimized stuff like the radix-4 FFT and multitone signal synthesis. You would also need the sound card and the ADI development tools to do any work on it. I also have several programs that deal with Golay(24,12), Reed-Solomon, and BCH block codes.

Phil promised that he would be making his convolutional coding program available after doing some cleanup work. I am quite interested in trying that with the OFDM modem.

I also suggest looking at Eric's OFDM simulations - it is very enlightening.

73's

Johan, KC7WW

36. Re: [HFSIG:207] Re: HF Modulation

From: Greg Jones <gjones@tenet.edu>
Date: Wed, 18 Jan 1995 16:27:56 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199501182227.QAA09979@Alice-Thurman.tenet.edu>
Rendered body type: text/plain

According to Johan Forrer

FL:

> Thanks for bringing this up - Greg suggested we use tcet.unt.edu in
> the mean time and I checked and found that there indeed was a TAPR
> subdirectory. I am not sure what the future plans are for a permanent home
> for these code fragments.

ftp.tapr.org should be registered as a mx record later today and should be available for access shortly, once the it distributed fully (figure a few days).

HF SIG will have a location for this type of thing. I'll let you know when we I think it is all operational. Johan will get a special login so that you can manage the area.

/tapr/SIG/hfsig

Should be better than TCET :-)

Greg

37. Re: HF modulation

From: bm@lynx.ve3jf.ampr.org (Barry McLarnon VE3JF)
Date: Thu, 19 Jan 1995 02:07:56 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <m0rUmI3-0002hbC@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

[I sent this a few days ago, but it seems to have vanished in cyberspace... I guess because of the listserver changeover. Try, try again...]

I only tuned into this mailing list quite recently, so I'm just starting to catch up on some of the discussions. This message caught my eye:

> Phil's intriguing proposal for using M-ary orthogonal FSK
> with Walsh functions. Over Christmas break (I'm pursuing
> an MSEE; if only I could catch it) I did some crude
> simulations of Phil's scheme using MATLAB.
>
> First, I generated a 64x64 Hadamard matrix (whose rows are
> Walsh functions). Then I generated 64 equal amplitude

> sinusoids, each separated by 32 Hz, from 512 Hz to 2528 Hz.
> I then picked a row at random from the Hadamard matrix to
> determine which sinusoids to sum together. I sampled the
> time sequence at 32 kHz and plotted it over a full period
> of the combined signal.
>
> I used several different Walsh functions to get a feel for
> the general behavior of these signals. Since I started with
> 64 tones, and only 1/2 are on for any particular Walsh
> function, I was combining 32 separate tones. The results
> were rather interesting, and very similar to Johan's.
>
> Thus wrote Johan, KC7WW:
>
> > As a way to learn more about how OFDM works for the
> > 39-tone MIL-STD-188 parallel modem, I wrote a C program
> > that simulates 39 parallel tones and combines them into
> > a single output signal.
>
> [snip]
>
> > I also did capture the simulated output waveform and was
> > quite concerned after inspecting the plot - the
> > presence of large spikes of significant magnitude (in
> > some cases, some 10-15 dB over the RMS value).
>
> My simulation produced signals with peak-to-RMS ratios of
> 9 dB, they were generally of low amplitude except where
> the sinusoids came into phase and produced a large peak,
> and they generally looked 'noiselike'. I also noticed that
> the signal's peak amplitude was not symetric about the time
> axis. The signals had zero average value but the largest
> positive-going peak was twice as high as the largest
> negative-going peak.

Are you sure about the 9 dB? A 32-tone signal should have a peak-to-RMS ratio (aka "Crest Factor") of 18 dB for the case in which all of the tones have zero initial phase shift. The formula for the Crest Factor for N tones is:

$CF = \sqrt{2N}$, or in dB, $20 \log CF$.

> As several others have noted, this high peak-to-RMS ratio
> will limit the amount of energy per symbol. A signal such
> as this is potential trouble in any amplifier unless care is
> taken to control the peak level. The asymetry in peak
> heights may further limit the amount of power an amplifier
> could deliver. This is definitely not a constant envelope
> signal so linear amplifiers are in order. As more tones are
> added the signals will have larger peak-to-RMS ratios. This
> is consistent with Johan's observations since he was summing
> more tones.
>
> The FCC limits amateurs based on PEAK envelope power (PEP),
> not RMS power. As Phil noted, coding gains can buy back
> most of the loss in energy per symbol caused by the low RMS
> value of the signal, but it's still a significant limitation.
> Disregarding interference, it's ultimately the energy per
> bit that determines the performance of a digital system.
>
> Phil's proposal was for a noncoherent system, which got me
> thinking (scary, eh?). A noncoherent system ignores the
> phase of a signal; so, we have a free parameter to play with.
> My initial simulation started each sinusoid at zero phase.
> If we could use the phase to reduce the peak value of the
> signal we could get more energy per symbol and get the coding
> gain as well.
>
> First I tried a linear shift in the initial phase across all
> 64 sinusoids. I started with zero initial phase at the lowest
> frequency and increased the phase linearly to two pi at the
> highest frequency. This resulted in signals with about the
> same peak-to-RMS ratios (8.25 dB), still 'noiselike', but the
> amplitude peaks became symetrical about the time axis
> (interesting and unexpected).

According to the reference I have (see end of this message), the CF remains at $\sqrt{2N}$ in the case of phase shifts which vary linearly with frequency, so it should still be 18 dB.

> My next attempt was to assign a random initial phase to
 > each of the 64 sinusoids. The initial phase was distributed
 > uniformly between zero and two pi. This produced signals with
 > an average peak-to-RMS ratio of 4.75 dB, the signals looked
 > extremely 'noiselike', and the amplitude peaks were nearly
 > symmetrical. Progress!

I think this is again off by a factor of two, but it illustrates that random phase is certainly a better choice... in this case, the dependency of CF on N is of the order of $\sqrt{\log N}$ instead of $\sqrt{N}$.

> These were just the first ways I thought of to change the
 > phase, there was no method to my madness. Even a naive
 > method like randomizing the phase produced a 4 dB reduction
 > in the peak-to-RMS ratio. These results lead to a conjecture
 > and several questions.

>
 > My conjecture is that we can reduce the peak-to-RMS ratio
 > further, and maybe even approximate a constant envelope
 > signal, by properly choosing the phases of the sinusoids.
 > I think this is plausible since a wideband FM signal is a
 > constant envelope signal and its spectrum looks like a set
 > of nearly equal amplitude sinusoids, within a limited
 > bandwidth. I doubt we could get a truly constant envelope
 > signal because we are using a small set of harmonically
 > related sinusoids.

Your conjecture is correct. :-)

> Some questions I have:
 >
 > - Are there systematic methods for choosing the
 > initial phases of the sinusoids to minimize the
 > peak-to-RMS ratio?

Yes. See the paper by Boyd referenced below. A proof exists that, in the case where N is a power of 2, a set of phases can be found which yields a Crest Factor of under 6 dB for arbitrarily large N. Moreover, he gives a procedure for selecting phases which yields a CF of about 4.6 dB for *any* N (except when N is very small). The waveform generated by this procedure is fascinating: it closely resembles a chirp signal (constant envelope swept frequency, sometimes used as a spread spectrum technique).

> - What limits are there on reducing the peak-to-RMS
 > ratio?

According to Boyd, there is evidence that the ultimate limit is around 3 dB, but no systematic method for finding the phases exists (yet). A colleague of mine did some work a few years back on numerical methods for finding near-optimum phases for multitone modems. He used an example of a 12-tone FSK modem (i.e., 12 simultaneous narrow-shift FSK signals with regular frequency spacing - one of the CCIR standards) and was able to find phase choices which yielded a CF in the 4-5 dB range for all 4096 possible frequency combinations. This would be a considerable improvement over the 13.8 dB CF you'd get with linear phases.

> - Are these limits, if they exist, related to the
 > number of sinusoids or the Walsh function used?

The limits seem to be independent of N, as long as N exceeds some minimum number (of the order of 10-20).

> - Is this a well-known (or obscure) problem that
 > already has a solution?

It's on the obscure side, but certainly some work has been done. I haven't been involved in HF data comms for quite a few years, so I don't know what recent advances might have been made in multitone modems... but I can ask around a bit.

> - Is reducing the peak-to-RMS ratio really as
 > important as I seem to think it is?

I'd say it's certainly worth further investigation... with the processing power now available in DSP's, control of the CF seems a lot more feasible than it used to be. Sometimes higher average power will buy you an improvement in BER, and sometimes it won't (when you've hit an

irreducible error rate due to intersymbol interference from multipath), but overall it would be a win.

> The MIL-STD modems apparently use PSK on the tones as well.
> Since the phase of any particular sinusoid in my simulation
> was constant over the symbol period this should allow using
> methods such as DPSK on the tones if someone wants to add an
> additional layer of modulation. Would this increase the
> peak-to-RMS ratio?

I'm pretty sure it would, since you're now putting an additional constraint on the phases. If you want to Keep It Simple, stick with noncoherent FSK (which generally works at least as well as DPSK in multitone HF modems anyway).

References:

S. Boyd, "Multitone signals with low crest factor", IEEE Trans. on Circuits and Systems, vol. CAS-33, no. 10, October 1986, pp. 1018-1022.

S. Shlien, "Minimization of the Peak Amplitude of a Waveform", _Signal Processing_, vol. 14, no. 1, January 1988, pp. 91-93.

> Eric, N2NNP

Barry

--

Barry McLarnon VE3JF/VA3TCP
Ottawa Amateur Radio Club Packet Working Group
Email: bm@hydra.carleton.ca or bm@lynx.ve3jf.ampr.org

38. Re: [HFSIG:209] Re: HF modulation

From: "Johan Forrer"
Date: Thu, 19 Jan 1995 09:38:03 PST8PDT
To: hfsig@tapr.org
Message-ID: <E219F550CBF@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Barry McLarnon VE3JF/VA3TCP wrote:

> [I sent this a few days ago, but it seems to have vanished in
> cyberspace... I guess because of the listserver changeover. Try, try
> again...]

This appears to happen occasionally - please just keep trying.

> I only tuned into this mailing list quite recently, so I'm just starting
> to catch up on some of the discussions. This message caught my eye:

<...lines deleted...>

Welcome aboard Barry - looking forward to your contributions.

73's

Johan, KC7WW

39. [HFSIG:209] Re: HF modulation

From: BLOOM_ALAN/HP5300_R0@opnmail3.corp.hp.com
Date: Thu, 19 Jan 95 11:04:00 -0800
To: hfsig@tapr.org
Message-ID: <d0eHfbY00000000000@MHS>
Rendered body type: text/plain

Barry McLarnon VE3JF (bm@lynx.ve3jf.ampr.org) wrote:

>> My simulation produced signals with peak-to-RMS ratios of
>> 9 dB, ...

>Are you sure about the 9 dB? A 32-tone signal should have a peak-to-RMS
>ratio (aka "Crest Factor") of 18 dB for the case in which all of the
>tones have zero initial phase shift. The formula for the Crest Factor for
>N tones is:

>CF = sqrt(2N), or in dB, 20 log CF.

Note that this is the ratio of the peak power at the top of the sine wave(s) to the total RMS power. The PEP to RMS ratio will be 3 dB lower.

Even a single pure sine wave has a peak-to-RMS ratio of 3 dB.

AL N1AL

40. 8ary FEK modem

From: chbrain@dircon.co.uk (Charles Brain)
Date: Thu, 19 Jan 1995 19:44:07 GMT
To: hfsig@tapr.org
Message-ID: <199501191944.AA24765@felix.dircon.co.uk>
Rendered body type: text/plain

Hi all,

Just b4 xmas I started trying to do a bit of DSP programming on a TMS320C50 DSK.

I thought I would try to write an 8 ary modem a la MIL-STD 188-141A. I am using a 10 Khz sample rate (mainly because all the ALE tones are multiples of 250Hz) and I use the sample rate to generate the 8ms pulse periods table lookups etc.

Well the Golay coding / triple redundancy and tone generation seems to work fine.

However my idea of using 8 iir bandpass filters seems to have been a total disaster! Mainly as I can only use 2nd order filters in the available time.

I had thought of using Goertzel's algorithm commonly used in DTMF decoder algorithms, but from what I have read 8ms doesn't seem to be long enough to decode the tones.

I could use an FFT I guess, but how do I get bit synchronisation? What ever method I use I need to get bit sync.

Anyone got any ideas on how best to synchronise and decode these tones?

Excuse my ignorance but this is the first DSP application I have tried to write.

By the way Phil thanks for the Specs, I got them last week, maybe after I get this working I will start on them next!!

Regards Charles G4GU0

"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)
Chelmsford, Essex.
E-mail chbrain@dircon.co.uk
POTS +44 (0)1245 353221
FAX +44 (0)1245 275448

41. Re: [HFSIG:212] 8ary FEK modem

From: "Johan Forrer
Date: Thu, 19 Jan 1995 19:00:26 PST8PDT
To: hfsig@tapr.org
Message-ID: <E2AFF881204@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Charles,

Well thats excellent work! I think you will find this an interesting excersise.

If you are looking for some good placed to read about this: Look at J.D. Ralph's book on Piccolo that was mentioned a while ago. There are a lot to gain about how to perform sync. Another good one in the paper by Chadwick and Springett in IEEE Trans. Comm Vol 18(#6) December 1970 " The design of a low data rate MFSK communication system". They mention two commonly used techniques.

Just a warning about the TI320C50 (applies to the 320C2x as well): beware of the REP instruction. If you use that in code that is of any substantial time, it can lock out interrupts. The idea in all this stuff is to plan to break your code into smaller chunks, some that can be performed in non-time critical and others that need to be time critical. Then write a fast, small real-time kernel to glue it all together.

It is often possible to operate a demodulator in non-coherent mode, then let your protocol analyser help you achive bit sync, and once you have that, close the loop to operate in synchnous mode. This is what we do with both .

Anyway, IIR filters are bad news for many reasons - unless you have good practical experience - better leave them alone.

I don't know whether this helps.

Good Luck and keep us posted.

73's Johan KC7WW

42. Re: [HFSIG:183] Re: HF modulation

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 19 Jan 1995 19:54:09 -0800
To: esilbaug@afit.af.mil
Cc: hfsig@tapr.org
Message-ID: <199501200354.TAA23822@servo.qualcomm.com>
Rendered body type: text/plain

Eric has demonstrated a fairly fundamental and powerful principle of statistics: the central limit theorem. That is, if you sum up enough independent random variables, eventually you get something that approximates a normally (Gaussian) distributed random variable.

It doesn't really matter what the input distributions are, although if the distributions are really non-gaussian it might take more of them to approach a gaussian sum.

The peak-to-average power ratio of a gaussian is of course infinite, so you have to pick some maximum level (corresponding to some number of standard deviations past the norm) and set your amplifier to clip there. If it's high enough, the lost energy will be negligible.

Phil

43. Walsh functions reference

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 19 Jan 1995 20:05:08 -0800
To: larry@owrlakh.unbc.edu
Cc: hfsig@tapr.org
Message-ID: <199501200405.UAA23827@servo.qualcomm.com>
Rendered body type: text/plain

I don't have any references offhand, but they're easy to describe.

A Hademard matrix:

is symmetric
is square
has a dimension that's a power of 2
is composed of only 0s and 1s
is defined recursively as follows:

$$H_{n+1} = \begin{bmatrix} H_n & H_n \\ H_n & \sim H_n \end{bmatrix}$$

with, by definition, $H_0 = |0|$

So if you apply this rule, you get

$$H_1 = \begin{bmatrix} 0 & 0 \\ 0 & 1 \end{bmatrix}$$
$$H_2 = \begin{bmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{bmatrix}$$

and so forth.

A Walsh function is simply a row (or column) taken from a Hademard matrix.

The interesting thing to note about the rows of a Hademard matrix is that they're mutually orthogonal. That is, if you pick any two rows at random, you'll find that the number of columns in which the two rows agree is equal to the number of columns in which they disagree.

Also, except for the first row, each row has an equal number of 0s and

1s.

One of the earliest uses for these things was to set up wire crossing plans for old open-wire telephone lines to minimize crosstalk. Very clever.

Phil

44. Re: [HFSIG:187] Re: G-TOR Booklet

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 19 Jan 1995 22:14:49 -0800
To: shall@rain.org
Cc: hfsig@tapr.org
Message-ID: <199501200614.WAA23928@servo.qualcomm.com>
Rendered body type: text/plain

>Rolf, Regarding G-TOR: I attended a talk given by Phil Anderson regarding
>G-TOR vs. Pactor and it appeared that G-TOR has much more powerful error
>correction than Pactor. I would be interested in your evaluation after
>you have had an opportunity to check it out. Steve, WM6P

Indeed it does. As I understand it, pactor has *no* error correction. All it does is accumulate bit energy across multiple transmissions of the same packet. This does increase E_b/N_0 and hence the chance of decoding a packet correctly, but it is much better to encode the data first to get some power gain as well as the E_b/N_0 increase.

To quantify things, simply sending the same packet twice and then combining transmissions increases E_b/N_0 by 3 dB. Encoding the packet in a strong rate 1/2 soft-decision-decoded convolutional code and then sending the result in two transmissions not only increases the E_b/N_0 by 3 dB, but also provides up to 7 or 8 dB of coding gain, for a total improvement of 10-11 dB.

G-TOR also uses a Type II Hybrid ARQ/FEC protocol, which I've been advocating for some time because of its ability to combine the best of FEC (ability to deal with rotten channels) with the best of ARQ (adaptability to rapidly changing channel conditions).

Unfortunately, G-TOR uses a relatively weak code (Golay) and as far as I know does hard decision decoding. So the coding gain isn't as great as it could be; I think it's only a couple of dB.

Phil

45. Re: [HFSIG:216] Re: G-TOR Booklet

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Fri, 20 Jan 95 8:23:20 MET
To: hfsig@tapr.org
Message-ID: <9501200723.AA04434@ife>
Rendered body type: text/plain

> ...As I understand it, pactor has *no* error
> correction. All it does is accumulate bit energy across multiple
> transmissions of the same packet.

Yes, this is correct; Pactor uses a simple Memory-ARQ protocol without any error-correction coding (i.e. it is not a hybrid protocol at all).

> G-TOR also uses a Type II Hybrid ARQ/FEC protocol, which I've been
> advocating for some time because of its ability to combine the best of
> FEC (ability to deal with rotten channels) with the best of ARQ
> (adaptability to rapidly changing channel conditions).

>

> Unfortunately, G-TOR uses a relatively weak code (Golay) and as far as
> I know does hard decision decoding. So the coding gain isn't as great
> as it could be; I think it's only a couple of dB.

I was involved in developing an HF-modem over the last 4 years or so. Unfortunately it never made it to the market :-(
It employed such an adaptive Memory Type II Hybrid ARQ/FEC protocol on top of a constant-envelope offset QPSK modulation. The memory-ARQ was augmented by a kind of Chase' code-combining using convolutional-codes for error-correction. Indeed, this modem allowed us to adapt over a wide range of signal-to-noise ratios and achieved still some bit/s throughput at an SNR of -20 dB (measured at 3 kHz bandwidth), whereas all other modems I tested stop working at around -9 dB. So this confirms Phil's estimate of a 10-11 dB gain.

The GTOR booklet should be on its way to HB9...

73,
Rolf
--

--- * * * ---

Rolf Sommerhalder	
Swiss Federal Institute of Technology (ETH)	home address:
Electronics Laboratory ETZ H60.1	c/o Rindlisbacher
Gloriastrasse 35	Wermatswilerstrasse 96
8092 Zurich / Switzerland	8610 Uster / Switzerland
Voice +41 1 632 76 40 (or 632 51 53)	Voice +41 1 941 59 28
Fax +41 1 632 12 10	AX.25 HB9CWP @ HB90S
E-Mail rolf.sommerhalder@ife.ee.ethz.ch	Swiss Disaster Relief HBK81

46. Re: [HFSIG:214] Re: HF modulation

From: "Johan Forrer"
Date: Fri, 20 Jan 1995 09:10:22 PST8PDT
To: hfsig@tapr.org
Message-ID: <E392A520687@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi All,

As I gain more working experience with these composite audio signals, I don't think we should, as Phil suggested, break our heads too much about the poor crest factor. In my 8-channel system, I scaled the RMS value way down so that the dynamic range of the D/A can handle the peaks. Playing the audio through the speaker amp, it sounds quite clean - nothing adverse to report at all. Actually, if one look at a sound clip on the Windows sound editor, it also shows some poor crest factor - though that's quite normal for everyday sounds.

Now here is the challenge for you technical types: Could someone (Al, Barry) explain what it would take in terms of linear amps etc, to be able to use this waveform, i.e. if one say have the rms level set at 1W output, and expect some 18 dB peaks, what should the RF amp have to deliver to pass the peaks. I know it's simple math, but just for everyone's curiosity. Thanks fellows.

73's

Johan, KC7WW

47. Hadamard matrices

From: Kok Chen <kchen@apple.com>
Date: Fri, 20 Jan 95 10:08:46 -0800
To: hfsig@tapr.org
Message-ID: <9501201808.AA08031@apple.com>
Rendered body type: text/plain

Just a small correction on Phil's posting. Hadamard is spelled with an "a" after the "d," not an "e."

Also, Hadamard matrices do not have to be a power of 2. The necessary condition is that the order is 2, or a multiple of 4. Not all multiples of 4 can exist. The first number where the condition is insufficient is order 188. See Berlekamp's Algebraic Coding Theory book for a discussion. (McGraw-Hill, 1968 - my copy predates the ISBN numbering, sorry).

One of the nice properties of Hadamard matrices which Phil did not mention is that the inverse of a Hadamard matrix is its own transpose scaled by the order of the matrix. This makes taking the inverse transform as easy as doing the forward transform.

(Hadamard is more well known for the "Hadamard Inequality" regarding the relationship between the determinant of a square matrix and the square magnitude of its elements.)

See Robert McEliece, "The Theory of Information and Coding," Addison Wesley, 1977, ISBN 0-201-13502-7 for a reference to the fact that the best block codes with a certain distance property are all related to the Hadamard matrix.

Finally, I'm not sure which came first - Rademacher-Walsh functions or Hadamard matrices. Both Walsh and Hadamard were contemporaries during

the early 1900s. I need to ask some mathematician friends about that.

73

Kok Chen, AA6TY
Apple Computer, Inc.

kchen@apple.com

48. Re: [HFSIG:217] Re: G-TOR Booklet

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 20 Jan 1995 14:35:24 -0800
To: hfsig@tapr.org
Message-ID: <199501202235.0AA03393@servo.qualcomm.com>
Rendered body type: text/plain

I just checked my references. Hard decision Extended Golay (24,12) over a BPSK or QPSK AWGN channel has a gain of about 2.5 dB at BER=1e-5. As expected, soft decision decoding buys another 2dB for a total of 4.5dB.

Convolutional codes can do much better at the same rate of 1/2. K=7 soft Viterbi decoded can give about 5.5-6.0 dB, and going to K=9 gives 6.0-6.5dB. (All at a BER of 1e-5). K=32 soft Fano (sequential) decoded gives about 7-7.5 dB of gain.

Phil

49. Re: [HFSIG:220] Re: G-TOR Booklet

From: "Johan Forrer
Date: Fri, 20 Jan 1995 17:01:40 PST8PDT
To: hfsig@tapr.org
Message-ID: <E41053C7F24@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Phil,

Those figures for coding gain on Golay that you mentioned, does it account for interleaving? In the Pierce paper they show that an effective constraint factor, i.e. high order of interleaving, in the order of 5000 - 50000 bits, used with the perfect codes (23,12,3) (Golay) and (127,64,10) works as well as any of the convolutional codes they tested, including concatenated convolutional - block codes. What am I missing - are these figures of merit apples and oranges? HELP!

Could you also perhaps give us some feedback on what kind of resolution would you need on the soft decision decoding (how many levels) would be practical, i.e. 16 bits would produce a very large trellis would'nt it? Is 4 levels sufficient?

Keep up the good work.

73's

Johan, KC7WW

50. Re: [HFSIG:221] Re: G-TOR Booklet

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 20 Jan 1995 18:42:24 -0800
To: hfsig@tapr.org
Message-ID: <199501210242.SAA03627@servo.qualcomm.com>
Rendered body type: text/plain

>Those figures for coding gain on Golay that you mentioned, does it account
>for interleaving? In the Pierce paper they show that an effective

Interleaving is a no-op on the AWGN channel. The error distribution is already uniformly random, so interleaving doesn't make any difference.

Of course, HF is *not* an AWGN channel. So interleaving can make a big difference there, as it does on the mobile fading channel.

Also, when you concatenate codes the channel seen by the outer code is decidedly non-AWGN even if the physical channel seen by the inner code is AWGN. For example, Voyager used Reed-Solomon over Viterbi, and Viterbi decoding errors generally occur in characteristic bursts until the decoder eventually resettles on the correct path through the trellis. Interleaving spread out these bursts over multiple RS blocks, making them easier to correct.

51. Eval copy of parallel modem

From: "Johan Forrer
Date: Sat, 21 Jan 1995 13:32:50 PST8PDT
To: HFSIG@tapr.org
Message-ID: <E558AE51424@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi folks,

I will place an evaluation copy of the parallel modem on the net on monday. The modem runs in real time on a PSA sound card - I have tested it on my Orchid Soundwave32 and will also test it on a Cardinal card over the weekend.

This is only a test setup but I am sure you will find it interesting. The PC host sends a test message over the 8-channel OFDM modem that runs on the DSP on the sound card and plays the demodulated message as received from the modem, on the screen. You will hear the tones going over the speaker and see the rate that it writes to the screen. This version occupies only 250 Hz, so you could multiply things by four to get a feeling of what is coming, of course it will be slower once we add the error-correcting code.

The next task is to work on the synchroniser and then the data-link layer. Then we will be able to start testing over the air.

I don't have FTP capability here from home, else would have put it on the net already. Look for the code monday at ftp.ucsd.edu in /hamradio/packet/tcpip/incoming/pmodem1.zip
I will include a short readme to help you set it up and run it.

73's

Johan, KC7WW

52. Re: [HFSIG:221] Re: G-TOR Booklet

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Mon, 23 Jan 95 8:09:59 MET
To: hfsig@tapr.org
Message-ID: <9501230710.AA05321@ife>
Rendered body type: text/plain

Johan wrote:

> Could you also perhaps give us some feedback on what kind of resolution
> would you need on the soft decision decoding (how many levels) would be
> practical, i.e. 16 bits would produce a very large trellis wouldn't it? Is
> 4 levels sufficient?

I should dig out a reference about this question which provides you with exact graphs how much you lose/gain by decreasing/increasing the number of quantization levels.

My own experience is that 256 level quantization (i.e. 8 bit) is fine, but 8 level quantization still works and the losses (versus 256 level quantization) are around 1/8 to 1/4 dB (if I recall those figures correctly). However, the performance of a 8 level quantized Viterbi decoder is heavily dependent on the correct setting of the quantization thresholds/levels, i.e. a perfectly adapted AGC is a must. The more quantization levels you spend in the decoder, the looser your gain control may be.

73,
Rolf

P.S. Most of the commercial Viterbi decoder chips available work with 3-bit (plus additional erasure symbol) or 3.5- or 4-bit quantization.

--

--- * * * ---

Rolf Sommerhalder	home address:
Swiss Federal Institute of Technology (ETH)	c/o Rindlisbacher
Electronics Laboratory ETZ H60.1	Wermatswilerstrasse 96
Gloriastrasse 35	8610 Uster / Switzerland
8092 Zurich / Switzerland	

Voice +41 1 632 76 40 (or 632 51 53)

Voice +41 1 941 59 28

53. Re: [HFSIG:217] Re: G-TOR Booklet

From: "Johan Forrer"
Date: Mon, 23 Jan 1995 09:00:23 PST8PDT
To: hfsig@tapr.org
Message-ID: <E8101787ACB@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Rolf Sommerhalder wrote:

<...lines deleted...>

> I was involved in developing an HF-modem over the last 4 years
> or so. Unfortunately it never made it to the market :-(
> It employed such an adaptive Memory Type II Hybrid ARQ/FEC protocol on
top of
> a constant-envelope offset QPSK modulation. The memory-ARQ was augmented
by
> a kind of Chase' code-combining using convolutional-codes for error-
correction.
> Indeed, this modem allowed us to adapt over a wide range of signal-to-
noise
> ratios and achieved still some bit/s throughput at an SNR of -20 dB
(measured at
> 3 kHz bandwidth), whereas all other modems I tested stop working at
around -9 dB.
> So this confirms Phil's estimate of a 10-11 dB gain.

Sounds like a nice accomplishment - congratulations! Have you though
about publishing something about it? I am sure we all would be very
interested learning more about it. OQPSK has a very nice and clean
waveform - I will look into that a little more. What kind of DSP is
this work based on?

I have been using 16-bit soft-decoding Memory-ARQ on the sound card
implementation of Pactor and have found it to help quite a bit - a bit
better than nothing at all.

73's

Johan, KC7WW

54. Re: [HFSIG:223] Eval copy of parallel modem

From: "Johan Forrer"
Date: Mon, 23 Jan 1995 09:04:52 PST8PDT
To: hfsig@tapr.org
Message-ID: <E81152C3BC6@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi all,

The files are now on ftp.ucsd.edu.

I would like to thank OM Kok Chen, AA6TY for the wonderful
discussions we have had on how to make this parallel modem work.

73's

Johan, KC7WW

55. RE: HF Modulation

From: Eric E Silbaugh <esilbaugh@afit.af.mil>
Date: Mon, 23 Jan 1995 17:53:53 -0500
To: bm@lynx.ve3jf.ampr.org
Cc: hfsig@tapr.org
Message-ID: <199501232253.AA28813@stealth.afit.af.mil>
Rendered body type: text/plain

Barry,

Thanks for the references! They just added to my depression :)
Seriously, it's nice to see that my conjecture wasn't too far
off the mark. They answer some of the questions I asked and
confirm that the problem is difficult.

In my simulations I was thinking in terms of power not voltage so

But I don't want to give up that 2dB without a fight. Getting soft decision decoding to work really well in my protocol, especially with non-Gaussian noise, may take some doing.

57. RE: HF Modulation

Barry,

In my simulations I was thinking in terms of power not voltage so I used $10 \cdot \log()$ not $20 \cdot \log()$. This is the source of the factor of 2 difference in our crest factor calculations. I probably should have been more specific; others were confused by this as well.

It was encouraging to see that limits on crest factor reduction are nearly independent of the number of sinusoids. This means the Walsh functions will control how well a particular set of phases works. I'm going to try the quadratic initial phase scheme from the Boyd paper and see what it does.

Thanks again for the answers, now I've got even more questions!

Eric, N2NNP

[illegible]

58. Re: [HFSIG:223] Eval copy of parallel modem

On Sat, 21 Jan 95 15:38 CST, "Johan Forrer FL" <FORRERJ@frl.orst.edu> wrote:

Johan, and All...

I FTPed the file and ran it and all I can say is keep going... It looks (and sounds) good... One note... The SW32 has a load feature that allows you to load the driver... Just do a "SW32 /F:parll.ld" and then run MODEM...

Johan, what are the clicks... I distinctly hear tones but also some low frequency "clicks"... Us novices want to know... ;-)

--
Marv...

-- Marv Uphaus -- muphaus@cris.com -- Ph: 334-343-9256 --
-- U.S.Mail: 4031 Airport Blvd. #49 -- Mobile, AL 36608 --
-- Packet Radio Address -- K4BVG @W4IAX.#MOBAL.AL.USA.NA --

59. Crest Factor Control

From: Eric E Silbaugh <esilbaug@afit.af.mil>
Date: Wed, 25 Jan 1995 16:20:39 -0500
To: hfsig@tapr.org
Cc: esilbaug@afit.af.mil
Message-ID: <199501252120.AA14845@stealth.afit.af.mil>
Rendered body type: text/plain

This is a toothpaste commercial, NOT!

WHY CREST FACTOR CONTROL

Thus wrote Johan, KC7WW:

> Actually, if one look at a sound clip on the Windows sound editor, it
> also shows some poor crest factor - though thats quite normal for
> everyday sounds.

Speech is well known as a 'peaky' (high crest factor) signal, thus the voice processors (peak clipping) used by DXers. The other extreme is RTTY overheating a linear amp; higher average power.

Thus wrote Barry, VE3JF:

>> - Is reducing the peak-to-RMS ratio really as
>> important as I seem to think it is?
>
> I'd say it's certainly worth further investigation... with the
> processing power now available in DSP's, control of the CF seems a lot
> more feasible than it used to be. Sometimes higher average power will
> buy you an improvement in BER, and sometimes it won't (when you've hit
> an irreducible error rate due to intersymbol interference from
> multipath), but overall it would be a win.

Johan also enscribed:

> As I gain more working experince with these composite audio signals, I
> don't think we should, as Phil suggested, break our heads too much
> about the poor crest factor.

I have to agree. High crest factors are not a 'show stopper' but will make using multitone modems more difficult. Likewise, low crest factors will lower the peak instantaneous power which should reduce interference; they are not a cure-all.

If we control the crest factor we can get more energy per symbol, for the same input signal level. Low crest factor signals will also reduce the chances of accidentally overdriving an amplifier. This would make the modem easier to use, which would be a real benefit. How much more energy and how much easier to use are open questions.

We need not worry about getting the absolute lowest crest factor; instead, let's concentrate on controlling the crest factor to a reasonable value. In the paper Barry mentioned, Boyd was developing multitone test equipment and notes that crest factors of 6 dB are acceptable for many applications. We can get a basic proof-of-concept modem running, begin testing, and then refine it.

WHERE TO GO FROM HERE

Looking towards an implementation, if we stay with incoherent demodulation we may not have to standardize which phases to use. Just choose any set of phases giving reasonable crest factors; the magnitude spectrum will be unchanged. It would be good to provide a standard set so not everyone has to go through the minimization process. Who knows, there may be patterns which could lead to an empirical algorithm for choosing a good set of phases; if not, we could use a look-up table to set the initial phases.

SIMULATIONS

Using $20 \cdot \log(\text{peak}/\text{RMS})$, the crest factor for the zero phase case is 18 dB. With quadratic initial phase the crest factor is between 7-8 dB. Simulation of 20 different signals gave an average of 7.52 dB. That's 10 dB less than the zero phase case and may be good enough! Many of the signals looked like chirp signals, just like the examples in the Boyd paper. In a few cases my naive random initial phase method produced a slightly lower crest factor. This was very rare; in most cases the random method was 1-2 dB higher.

Multitone signaling, OFDM, is one of several solutions we should evaluate. From a system view it has some attractive features: multiple tones combat selective fading caused by multipath, and M-ary signaling reduces the effects of inter-symbol interference. Channel coding to correct errors and automatic power control will probably also be part of any robust, low interference HF modem. However, we can start simple and add complexity only when necessary.

Eric E. Silbaugh
esilbaug@afit.af.mil

All standard, non-standard, and MIL-STD disclaimers apply

From: "Jonan Forrer"
Date: Mon, 30 Jan 1995 09:17:47 PST8PDT
To: hfsig@tapr.org
Message-ID: <8A5E5867A@frl.orst.edu>
Rendered body type: text/plain

Eric Silbaugh wrote:

Page 139

> set the initial phases.

Good suggestion Eric - this looks like the way we should do it.

I only wish that there was some solution for QPSK. It appears that offset PSK is one possibility, perhaps waveshaping similar to generate MSK waveforms - may help to produce a constant envelope? I will be playing with that idea a bit more when I get a bit of time. Does anyone know how this is done on the Harris 39-tone modems?

73's

Johan - KC7WW

61. Packet software for sound card

From: "Johan Forrer
Date: Mon, 30 Jan 1995 16:02:15 PST8PDT
To: HFSIG@tapr.org
Message-ID: <911C7750DB@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi All,

This does not really belong in the HFSIG - just for those interested.

I have put the HB9JNX sound card modem software on the "upload" directory. This is Thomas Sailer's 1200 and 9600 (G3RUH compatible) DSP modems for PSA sound cards. Assuming you know how to integrate such software into one of the "NOS" flavors, things will make sense.

Tom also has a neat RTTY/AMTOR/Pactor package that uses the sound card for his DSP modem - scalable tones, FFT display etc. I will upload that as well.

Thanks to Tom for making his software available.

73's

Johan, KC7WW

62. Re: [HFSIG:233] Packet software for sound card

From: muphaus@cris.com (Marv Uphaus)
Date: Mon, 30 Jan 1995 22:24:34 -0600
To: hfsig@tapr.org
Message-ID: <2kRBLcysSgj3077yn@cris.com>
Rendered body type: text/plain

On Mon, 30 Jan 95 18:06 CST, "Johan Forrer

FL" <FORRERJ@frl.orst.edu>

wrote:

>Tom also has a neat RTTY/AMTOR/Pactor package that uses the sound
>card for his DSP modem - scalable tones, FFT display etc. I will upload
>that as well.

I got Tom's software last week, Johan, and it seems to work quite well... What it doesn't do is reset the PSA card back properly... Nothing I could do would set the PSA card back properly for WinDOZ... It wouldn't reload the VSW32.386 driver properly... Only turning the computer off and back on would reset the card...

Tom, are you reading this...??? Anybody know a fix for this or is it a coding problem in the exit from the DSP software... I do, of course, know to reload the GENMID.LD driver... That seemed to go OK, but WinDOZ would never load the VSW32.386 driver until the computer was turned off and on again...

>Thanks to Tom for making his software available.

You bet, Tom... Nice...!!!

--
Marv...

-- Marv Uphaus -- muphaus@cris.com -- Ph: 334-343-9256 --
-- U.S.Mail: 4031 Airport Blvd. #49 -- Mobile, AL 36608 --
-- Packet Radio Address -- K4BVG @W4IAX.#MOBAL.AL.USA.NA --

63. Re: [HFSIG:234] Re: Packet software for sound card

From: "Johan Forrer"
Date: Tue, 31 Jan 1995 08:40:24 PST8PDT
To: hfsig@tapr.org
Message-ID: <A1BF76523B@frl.orst.edu>
Rendered body type: text/plain

FL" <FORRERJ@frl.orst.edu>

Hi Marv,

I have E-mailed to Tom and will post the new version in the upload area.
Will let you know.

For those interested, I will post my spectral display program for the PSA
cards in the upload area. It works, though is nothing fancy - just a
passing milestone during the OFDM research. look for FFTSCOP.ZIP.

73's

Johan, KC7WW

February 1995

Original source file: hfsig.9502

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1995-02/TAPR-HF_1995-02.txt

1. Parallel tone crest factor

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Fri, 3 Feb 1995 08:51:39 PST8PDT
To: HFSIG@tapr.org
Message-ID: <E9F1E470AD@frl.orst.edu>
Rendered body type: text/plain

Hi All,

I thought that you might find it interesting that the 39 tone parallel tone modems (MIL-STD-188-110) uses the Newman phase sequences. I calculated those and compared it to those listed in the MIL-STD docs and they are essentially the same. Then plugged the Newman values for initial phases into my simulation program and found that the crest factor is less than 6 dB, typically 5 dB over a signaling element. The signal also has that apparent chirp-like character. Quite exciting to see this magig work.

I then did the same thing on the DSP implementation and found that one cannot actually tell the difference by ear! Sounds pretty much the same as a random initial sequence.

73's

Johan, KC7WW

2. ARRL Digital Conference

From: "Rick Whiting" <Rick_Whiting@ATK.COM>
Date: 3 Feb 1995 17:16:11 -0600
To: "Post hfsig TAPR" <hfsig@tapr.org>, "Post Netsig TAPR" <netsig@tapr.org>
Cc: "Dave Bushong" <dbushong@wang.com>, "Kay Craigie" <4087290@mcimail.com>,
"Cornell Drentea" <cdrentea@AOL.COM>, "Gary Garriott" <garyg@vita.org>,
"David Henderson" <N3EOY^aN3EOY@p10.f165.n109.z1.fidonet.org>,
"Craig McCartney" <73126.3260@compuserve.com>,
"Bo McLean" <487-1998@mcimail.com>,
"Richard Muffley" <rmuffley@vita.org>,
"All Participants NETF" <netf@jriver.com>,
"Paul Newland" <paul.newland@att.com>,
"Vic Poor" <71151.720@compuserve.com>,
"Eric Rosenberg" <ericro@vita.org>,
"Dale Sinner" <73074.435@compuserve.com>,
"David Speltz" <5571530@mcimail.com>
Message-ID: <199502032311.RAA27227@ATK.COM>
Rendered body type: text/plain

Subject: OFFICE MEMO Time:5:07 PM
ARRL Digital Conference Date:2/3/95

Announcement and First Call For Papers

ARRL DIGITAL COMMUNICATIONS CONFERENCE
September 8-10
Arlington, Texas, U.S.A.

Mark your calendar and start making plans to attend the year's premier event in digital communications. The 14th Annual ARRL Digital Communications Conference will be held September 8-10, 1995, in Arlington, Texas -- just minutes from the Dallas/Ft. Worth Airport. Co-hosts for the Conference are TAPR and the Texas Packet Radio Society.

The ARRL Digital Communications Conference is an international forum for radio amateurs and experts in digital communications, networking, and related technologies to meet, publish their work, and present new ideas and techniques for discussion. Presenters and attendees will have the opportunity to exchange ideas and learn about recent hardware and software advances, theories, experimental results, and practical applications.

Anyone interested in digital communications is invited to submit a paper for publication in the Conference Proceedings. Presentation at the Conference is not required for publication. Papers are due by July 21, 1995, and should be submitted to Maty Weinberg, ARRL, 225 Main St., Newington, CT 06111 U.S.A. or via Internet at lweinberg@arrl.org. Please contact Maty for detailed format requirements.

The hotel/conference center is extremely well suited and well sited for this event. They regularly host other communications-related events and staff members are very familiar with the sight of satellite dishes springing up on the lawn! They and we are looking forward to a tremendous conference. We hope YOU can be part of it!

For more information on the Conference, registration, and hotel reservations please contact TAPR at the address below:

TAPR
8987-309 E. Tanque Verde Rd. #337
Tucson, AZ 85749-9399, U.S.A.
Phone: (817) 383-0000
Fax: (817) 566-2544
Internet: tapr@tapr.org

[2-3-95]

3. Piccolo etc.

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Tue, 14 Feb 1995 10:00:43 +0100 (MET)
To: HFSIG TAPR <hfsig@tapr.org>
Message-ID: <Pine.ULT.3.91.950214095523.9344B-100000@dxcern.cern.ch>
Rendered body type: text/plain

Hi all,

I just want to distribute a message I got from Adrian Nash, G4ZHZ who has written DSP code for the various Piccolo modes. I am hoping to get him to join the list as I think he has something to contribute. Perhaps some of you would like to contact him and encourage him to join HFSIG. I have spoken to Johan about the possibility of porting Adrian's code to the PSA sound cards like Cardinal.

Here is Adrian's message:

>From nash@camail.ca.nmp.nokia.comTue Feb 14 09:21:45 1995
Date: Mon, 13 Feb 1995 18:16:30 GMT
From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
To: frode@dxcern.cern.ch
Cc: nash@camail.ca.nmp.nokia.com
Subject: RE: G-TOR & other modes

Hello Frode

Yes, it would be good to get some amateur piccolo going. The software is written for the ADSP-2101 and requires 2k program memory and 1k data memory. This means it will run on an ADSP-2101 without additional off-chip memory. There are several versions:

ITA52 - 12-tone MFSK 110baud ASCII ITA5
ITA21 - 6-tone MFSK, Baudot 75 baud as used by UK government (but encrypted of course).
Also, experimental 32-tone Baudot piccolo - the original 1962 MKIII version which uses 32 different tones 10Hz apart for the 32 characters of the ITA2 code. This is really the rolls-royce of piccolo systems giving ultimate performance at the expense of bandwidth.

The modes are all in one 27C512 boot ROM.

So far, I have only laboratory tested the ITA5 mode but have found that it works very well - 1% bit-error-rate at -10dB SNR - ie you cannot hear the signal for the noise. This is of course without error correction. This is about 100 times better than a perfect 110bps synchronous FSK link at the same SNR in an AWGN channel.

The software is full-duplex and is currently designed to run on my own DSP modem. However, it could very easily be adapted to run on another system. I do not have the Cardinal PC sound card that you mention. How much and where could I get one from?

I intend to improve the performance and port the software to the TMS320C50 TI DSP starter kit.

The main difference between the way the commercial LA-1117 analogue MFSK modem works (no longer used by HM Government) and my modem is that the synchronisation is modulation-derived. This means that the modem will

accurately acquire synchronisation upon a MFSK signal which is sending data rather than requiring a period of idle (alternate 500/520Hz tone) signal. This makes it much easier for amateur use.

The only problem I foresee, is that the tuning must be very accurate. However most modern radios have at least a 10Hz synthesizer resolution. My modem has a form of visual tuning aid which helps. However, I may develop an automatic tuning system which acquires the signal using a form of digitally derived AFC.

I would like to hear more about the other modems you mention, 16 tone and 39 tone. Has anyone tried the Kineplex type modes which use DPSK and get 4800bps through a 3kHz bandwidth?

My MFSK modem can have a data-mode which uses 16 tones to transmit 4-bit binary words. With the standard 50ms element length, it will transmit at 80bps. Channels can be multiplexed up to occupy an SSB bandwidth. A rough calculation suggests that 1200bps can then be sent using 15 MFSK channels in a 3kHz bandwidth. This would allow packets to be sent over very poor HF channels since MFSK is very good in weak signal/interference and selective fading. I think that by applying some error correction (eg CRC) and interleaving (to overcome static crashes) we could provide 1200bps packet links on HF which have a very good throughput to rival that of pactor and clover.

If you think other people would be interested in MFSK, please forward my email as appropriate (eg TAPR).

73, Best Regards

Adrian G4ZHZ
Tel: +44 276 419430
Fax: +44 276 419260

Nokia Mobile Phones (UK) Ltd
Ashwood House
Pembroke Broadway
Camberley
Surrey GU15 3SP
England

73 Frode, F/LA2RL

```
*****
*      Frode Weierud           Phone   :      41 22 7674794      *
*      CERN, SL                Fax     :      41 22 7679185      *
*      CH-1211 Geneva 23       E-mail  :      frode@dxcern.cern.ch *
*      Switzerland            *
*****
```

4. HFSig Status

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Tue, 14 Feb 95 10:30:06 nst
To: hfsig@tapr.org
Message-ID: <9501147927.AA792786812@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

Hi,

I have only just signed onto the HFSIG list.

I have been reviewing the archives of hfsig, its going to take awhile...(there lots - almost 3" of printouts)

I'm interested in the current status of the new modem, the HF simulator, and the like. Has anyone a summary of the results to date?

I have been working on a project with a "4 of 1of4 tone FSK", i.e. 4 tones of 16, with an FFT based receiver. Peak amplitude optimization, receiver synchronization, protocols for a synchronized scanning system, error correction, etc. Preliminary on air testing shows the concept works, but needs improvement. Currently 125baud, 4 dibit channels = 1000bps.

I saw the early notes on peak optimization, and have achieved voltage gains of *1.7 to *2.3 on the four tones by precalculating the starting phases for all the possible tone combinations and keeping them in a lookup table. More info available if anyone is interested. Or better yet do you have better results???

My hardware is currently TMS320E17 based, with a TMS320C31 design in the works. The latter exists only as a bench grade item at present.

I'm running over Harris RF3200 and Skanti 8000 Radios (nice stuff, belongs to the company, glad I didn't have to pay for it), and several other brands of Marine HF SSB Radios.

About Me:

I'm primarily working on this for commercial reasons - its my job, but am open to working in the experimental and development effort, sharing test results and experiment time. Our main work is in protocols more so than modems, used for ship-shore data reporting here in Canada, and helicopter flight following (for Search and Rescue) in places like Rwanda and Cambodia. But like everybody else, My boss and our customers want higher throughput, but extremely robust at the same time.

Paul G Russell
St. John's, Newfoundland, Canada
paulr@neweast.ca
Fax: Canada (709)576-2125

"Engineering is a Profession in laziness - How to get the most done with the least amount of effort" - PGR

"You have got to be half crazy, otherwise all the nuts in the world will drive you totally insane" - PGR
corollary: "Which one am I?"

"Oops, Shouldn't have done that..." - PGR et al

Oh yea, my opinions are mine, and maybe others as well?

5. Re: [HFSIG:238] Piccolo etc.

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 14 Feb 1995 15:17:13 PST8PDT
To: hfsig@tapr.org
Message-ID: <1F8662D5415@frl.orst.edu>
Rendered body type: text/plain

Frode Weierud <frode@dxcern.cern.ch> wrote:

> I just want to distribute a message I got from Adrian Nash, G4ZHZ who has
> written DSP code for the various Piccolo modes. I am hoping to get him to
> join the list as I think he has something to contribute. Perhaps some of
> you would like to contact him and encourage him to join HFSIG. I have
> spoken to Johan about the possibility of porting Adrian's code to the
> PSA sound cards like Cardinal.

< lines deleted >

Just to let everyone know - there is a closeout sale for the Cardinal Pro 16+ (includes a SCSI 2 CD-ROM interface) at the moment - \$39.95 from J&R Music. These may not last long. I suspect everyone is getting ready for the new generation of DSP-based sound cards.

I think we all are interested in Adrian's work with MFSK. If he is interested, I for one, would be happy to help with the porting of the code to the sound card. Let me know if I can help.

73's

Johan - KC7WW

6. Re: [HFSIG:239] HFSig Status

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 14 Feb 1995 15:25:24 PST8PDT
To: hfsig@tapr.org
Message-ID: <1F889663FBA@frl.orst.edu>
Rendered body type: text/plain

Hi Paul,

Welcome to the group. I think that we all share a lot in common. You certainly have a lot of good experiences that we would like to

hear about. I can appreciate your business interests too - be sure to look after those. You will find that most of our work have been to provide a platform for experimentors and to advance the technology. I would hope that someone would find some of our efforts of use and go on to develop products for the amateurs (and commercial for that matter).

Paul Russel wrote:

> I'm interested in the current status of the new modem, the HF
> simulator, and the like. Has anyone a summary of the results to date?

I'll attempt to update you briefly on the present status:

SIMULATOR:

I think you will find the main literature references in the archives. We had good participation in this discussion that helped find and developed necessary principles to accomplish the task.

In order to test out some of these ideas, I wrote some "C" code for the Watterson model. This excersise also explored further DSP implementation issues. On the DSP side, I have completed coding and testing parts of the code, mostly the Hilbert transformer. There still remains a great deal of DSP code to be done. Most of it is relatively easy, that is if you have done it before, i.e. the random low frequency modulation effects need to be upsampled by a factor roughly 100:1. This requires typical multirate techniques, preferably with dithering. The random generator also needs a table-lookup algorithm. Just have not yet gotten an opportunity completing it all, but thought that the modem work needed more attention - I'll get back to it in all seriousness soon.

BTW, all the DSP work for the simultor that I have done so far, is for the ADSP 21xx, though would not take much to port to the TAPR DSP-93. I have good hope that it will eventually run on our present DSP platforms.

HF MODEM

There are several very good ideas floating around, and lately found that others actually may have some of it implemented.

Personnally, I have been experimenting with the OFDM idea, i.e. parallel-tone approach. My program uses 8/8 channels, each channel being QDPSK, symbol rate being 31.24 Hz ($2 \times 8 \times 31.25 = 500$ bps raw) in a 250 Hz channel. It uses a FFT-based demodulator. The idea is, DSP resources permitting, to expand to 32 channels (1000 bps raw rate). A rate 1/2 ECC will be added. In addition some form of ARQ data-link layer is being considered.

If you are interested - a demo version for this modem running on the sound card is available from the TAPR file server: [ftp.tapr.org](ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/parmdm1.zip)
[tapr/SIG/hfsig/upload/parmdm1.zip](ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/parmdm1.zip)

>
> I have been working on a project with a "4 of 1of4 tone FSK", i.e. 4
> tones of 16, with an FFT based receiver. Peak amplitude optimization,
> receiver synchronization, protocols for a synchronized scanning system,
> error correction, etc. Preliminary on air testing shows the concept
> works, but needs improvement. Currently 125baud, 4 dibit channels =
> 1000bps.
>

Sounds interesting - one of those ideas that we have seen here - just wonder about the baud rate, however. Have you had good experiences with that rate on the lower bands (40/80m) where multipath is so bad?

> I saw the early notes on peak optimization, and have achived voltage
> gains of *1.7 to *2.3 on the four tones by precalculating the
> starting phases for all the possible tone combinations and keeping
> them in a lookup table. More info available if anyone is interested.
> Or better yet do you have better results??? >

I assume you are talking about crest factor optimization? Have you tried

the Newman sequences - perhaps you have picked up some of our discussions and experimental results using that. My experiences for the 39-tone modem was that it is possible to get the crest factor down from 10+ dB to around 6 dB.

> My hardware is currently TMS320E17 based, with a TMS320C31 design in
> the works. The latter exists only as a bench grade item at present.
>

That is quite a big difference in capability - perhaps you interested in floating point capability? The C30 is a really nice chip - smart choice.

> I'm running over Harris RF3200 and Skanti 8000 Radios (nice stuff,
> belongs to the company, glad I didn't have to pay for it), and
> several other brands of Marine HF SSB Radios.
>

> About Me:
> I'm primarily working on this for commercial reasons - its my job,
> but am open to working in the experimental and development
> effort, sharing test results and experiment time. Our main work
> is in protocols more so than modems, used for ship-shore data
> reporting here in Canada, and
> helicopter flight following (for Search and Rescue) in places like
> Rwanda and Cambodia. But like everybody else, My boss and our
> customers want higher throughput, but extremely robust at the same
> time.
>

Sounds an interesting field, Paul.

>-----
> Paul G Russell
> St. John's, Newfoundland, Canada
> paulr@neweast.ca
> Fax: Canada (709)576-2125
>
> "Engineering is a Profession in laziness - How to get
> the most done with the least amount of effort" - PGR
>
> "You have got to be half crazy, otherwise all the nuts
> in the world will drive you totally insane" - PGR
> corollary: "Which one am I?"
>
> "Oops, Shouldn't have done that..." - PGR et al
>
> Oh yea, my opinions are mine, and maybe others as well?
>-----
>
>

73's

Johan, KC7WW

7. More Piccolo info from G4ZHZ

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Wed, 15 Feb 1995 09:38:53 +0100 (MET)
To: HFSIG TAPR <hfsig@tapr.org>
Message-ID: <Pine.ULT.3.91.950215093239.19866B-100000@dxcern.cern.ch>
Rendered body type: text/plain

I am posting yet another message from Adrian Nash, G4ZHZ.
I hope to get him to join the list in the next few days.

Here is a detailed description of his Piccolo modem:

>From nash@camail.ca.nmp.nokia.comWed Feb 15 08:47:55 1995
Date: Tue, 14 Feb 1995 18:21:10 GMT
From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
To: frode@dxcern.cern.ch
Cc: nash@camail.ca.nmp.nokia.com
Subject: RE: G-TOR & other modes

Hello Again

Thanks for your email. Yes, I am interested in a copy of the MIL-STD-188 spec.
I would be very grateful if you could send me a photocopy. I will look up for the Cardinal sound card and see if I can get one and adapt my software.

When I have done this, I will release the source code for the Cardinal Sound card. The problem is that I suspect that the 2115 is an ADSP-2105 with a host port added. This means that it will have 512 words of data RAM and 1k of program RAM. It may be necessary to "prune" the code to fit in this size.

I think the Analog Devices DSP architecture is very powerful and the assembler is very easy to learn for beginners. I would suggest that the ADSP-21xx is more powerful than the TMS320C50 or the DSP56002 since it has hardware for floating point manipulation, separate barrel shifter, multiplier and arithmetic units and ALL instructions take 1 cycle.

Here is a brief description of how my Piccolo modem works.

The modem is constructed on a eurocard using wire-wrapping. It uses a ADSP-2101 running at 12.288MHz. Other peripherals include an MC68681 DUART IC to perform serial comms and drive status LEDs. The ADC is a 12-bit AD7871 sampling at 7680kHz. The DAC is 8-bit. NB, a modern 14-bit CODEC chip is a far cheaper and a much more satisfactory solution but when I made the modem originally they were a bit expensive. The serial port supports ASCII at 9600 baud to connect to a PC or other device. It can be configured for 75baud Baudot (as used in ITA2 mode), then connect to a teletype.

The input signal is band-limited to 660Hz using a 3rd order Chebychev analogue filter. This is then sampled at 7680Hz by the ADC. The ADSP2101's on-chip timer is used to generate regular interrupts and drive the start-conversion pin on the ADC using the FLAG_OUT output.

The demodulator uses a 76-tap FIR filter to filter out excessive bandwidth. The bandlimited signal is decimated by 3 down to 2560Hz. This signal is then demodulated into the zero-IF complex plane using a quadrature NCO function. This function generates cosine/sine waves at 520Hz (the centre of the MFSK signalling band) using a 5th order Taylor series approximation. The complex signal is now in the range -80Hz to +80Hz corresponding to 380Hz to 680Hz inclusive. This includes the 12 MFSK tones 400Hz to 620Hz in 20Hz steps. The complex signal is now low-pass filtered using two cascaded FIR filters of 56 taps and 117 taps with a stage of decimation in between. The same LPFs are used for both I-phase and Q-phase. The bandwidth is a brick-wall 320Hz. The final output is decimated to 320Hz sample rate (NB: complex plane means we CAN sample at the Nyquist frequency).

The complex signal is converted to the frequency domain using a complex block-floating point, 16-point FFT algorithm. The FFT is a sliding FFT, that is a circular buffer is set up. Every sample which enters the buffer, the buffer moves along one (FIFO style) and the FFT is executed for EVERY sample. The 16 magnitude squared output correspond to a bank of 16 matched filters tuned to the 16 tones 20Hz apart. With the FFT being executed 320 times a second, this corresponds to 16 samples per 50ms MFSK symbol.

The modulation derived synchronisation (MDS) uses the following algorithm to extract a 20Hz sync signal:

$$C(n) = \frac{M(n)^2}{\sum[MAGk(n)]} \quad \text{where } k=0..15$$

ie C(n) the confidence estimate is a ratio of the strongest magnitude to the total power in the signal band (add up all the magnitude squared results). This gives a real-time estimate about how good the decision is going to be about which tone is the one transmitted. (Note this is in effect a soft decision so fans of Viterbi decoders as in V32bis modems may be interested!) After bandpass filtering this signal, a 20Hz signal results. A cross-correlator pattern-matches this to find the peak in the 20Hz signal which corresponds to peak confidence. This ensures that the decision is made at the optimum time. The pulses from the cross-correlator are used to lock a digital phase locked loop algorithm so that if more than one of the same tone frequency is transmitted (in which case the 20Hz signal is lost), the modem stays in sync.

The locked clock is used for triggering a decision. The selected tone is combined with the previous selected tone to index a look-up table which maps the ITA5 (12 tone) or ITA2 (6 tone) characters onto the tone pairs. The character is then checked for invalid combinations and idle character (500/520Hz). In order to prevent a false transition from the standby state to the traffic state, at least 9 idle characters one after the other must be received. However, there is a "break-in" switch which sets the modem running on a MFSK Tx which is already in progress. If the phase of the tones is wrong, another flick of the switch inverts the phase. (NB only a problem with the new 6/12 tone system where two tones are used - good old 32 tone piccolo only uses one tone so there is no ambiguity.)

On Tx, the timer interrupt handler reads characters from the UART and uses a lookup table to map the characters to tone pairs. These tone numbers are loaded into a FIFO buffer. The modulator uses a sine generation algorithm very similar to that used for the demodulator. The timer interrupt handler is actually called at 15360Hz and is time-shared between Rx tasks (eg counter updating and ADC conversion control) and Tx tasks (eg modulator).

The main-program runs two schedules: Rx control and Tx control. These call all the various components (eg demodulator, FFT, sync separation etc etc). The main program loop runs at the 320Hz final sample rate (ie 16 times per symbol). An Rx buffer buffers up enough 7680Hz input samples to allow the processing for one final sample.

The laboratory tests carried out in 1991/2 at University of Hertfordshire showed that in an Added White Gaussian Noise channel (AWGN), the modem performed extremely well. The BER was below 1 character error in 10E4 down to -5dB SNR. Below this the BER grows but is still only 1 character error in 10E2 at -10dB SNR. This is far superior to anything possible with normal FSK without any form of error correction. An EZ-LAB ADSP-2101 evaluation board was used as a test generator running a known pseduo-random sequence followed in the receiver. This allowed accurate BER measurements. On-air testing on a two way link has not been carried out yet.

However, qualitative evaluation using existing MFSK signals found on HF showed that although most of the time the encrypted signals could not be decoded, they were accurately demodulated with few errors being observed (ie no invalid tone combinations) in the presence of strong SSB phone, RTTY and broadcast stations. in-band CW tended to cause the most trouble. The modem would work on loop-back mode via a channel using a local transmitter and receiver. The main problem was actually trying to make the signal weak enough! Basically, 100% copy was obtained with a signal that was not even audible. This suggests that for normal QSOs, deep fading and interference are likely to have very little effect.

Future improvements

It was felt that improvements to the floating point arithmetic post-FFT would improve the dynamic range of the modem still further. More accurate sine generation in the demodulator would reduce the digital equivalent of reciprocal mixing which seems to occur with strong out-of-band signals at the ADC input. A better CODEC with more resolution would be more convenient though the greater resolution would do little more than represent the noise better unless the arithmetic and approximation quality of the algorithms can be improved.

For amateur radio use, some form of automatic signal acquisition would be useful. Although not too difficult to tune in accurately manually with a tuning aid, due to the variety of amateur HF radio equipment available, some form of AFC would help bring the modem on tune. An experimental algorithm has been tried particularly with regard to Doppler correction. However, at the time, it didn't work very well!

Full implementation of 16 tone mode. The demodulator uses a 16 point FFT. It is therefore just as easily capable of dealing with 16 tones. This would be used as a data mode in which 4 bits are conveyed using a tone. Two tones would convey an 8 bit word.

Original 32 tone piccolo. This system uses one tone for every element of the ITA2 alphabet. Each tone is 100ms long. The greater integration time available gives even better performance than the 6 dual-tone system. This system has been implemented partially in DSP. The modulation derived synchronisation technique removes the need for 10% AM modulation being applied to the MFSK signal as in the original UK FCO system. This enables the 32 tone mode to be used with any SSB radio equipment unmodified.

Well, I hope that has given you more insight into my MFSK activities Frode. Please post it up on the HFSIG list (by the way how do I get on that?) if you think people would be interested in having a go themselves.

73, es best regards

Adrian G4ZHZ

73 Frode, F/LA2RL

```

*      Frode Weierud           Phone   :      41 22 7674794      *
*      CERN, SL               Fax      :      41 22 7679185      *
*      CH-1211 Geneva 23      E-mail   :      frode@dxcern.cern.ch *
*      Switzerland            *
*****

```

8. Coding idea for a parallel carrier modem

```

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 15 Feb 1995 14:20 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <67C601F040C13121@chopin.ifj.edu.pl>
Rendered body type: text/plain

```

Just to stimulate the discussion I would like to say an idea I got recently about encoding bits and frames onto a multi carrier signals.

As a base I take the approach to have several regularly spaced carriers, each being modulated in phase.
 To decode/encode these signals we do "sliding window" FFT.
 Each window overlaps the one before and the one after by half of its size.
 At every window we change the carrier's phase by +90 or -90 degrees to encode 0s or 1s respectively. We are limited to 90 degrees shifts to keep adjacent symbols fully independent - this is because the windows overlap. Thus we encode one bit per FFT window.

Just to make things more concrete let's assume we take 64 samples per window at 9600 Hz sampling rate. This gives up 32 carriers spaced by 150 Hz. We only use 15 carriers: the first at 600 Hz and the last at 2700 Hz to fill up the 300-3000 Hz spectrum.
 There are 300 FFT windows/second which gives up 300 bits/second/carrier.
 Total raw bit rate would be $300 * 15 = 4500$ bps.

Encoding approach #0
 =====

Let's take the bits from say the HDLC encoder:

abcdefghijklmnopqrstuvwxy....

and code them straight onto the 15 carriers - like this:

```

      1 2 3 .... <- window number
1    a p .
2    b q .
3    c r .
4    d s .
5    e t .
6    f u .
7    g v .
8    h w .
9    i x .
10   j y .
11   k z .
12   l . .
13   m . .
14   n . .
15   o . .

^
carrier number

```

The receiver (after achieving the bit synchronization) would take the FFT of every window, compare the phases with the previous window, decode 15 bits and feed them (with a predetermined order) into the HDLC decoder. This approach gives us the full data bandwidth available but no protection against bit flips made by noise.

Encoding approach #1 (We make up a simple FEC)
 =====

We only send 11 data bits per FFT window and the remaining 4 bits we use for a simple FEC.

```

      1 2 3 .... <- window number
1    a l w
2    b m x
3    c n y

```

```

4   d o z
5   e p .
6   f q .
7   g r .
8   h s .
9   i t .
10  j u .
11  k v .
12  A E .
13  B F .
14  C G .
15  D H .

```

```

^
carrier number

```

ABCDEFGH... are the CRC bits designed to correct single bit flips.

My (possibly naive) way to compute the CRC bits would be:

```

x x x      x x x  x  A
x      x x  x x  x x B
  x   x  x x  x x x C
    x   x x  x x x x D
                        ^ CRC bits
a b c d e f g h i j k <- data bits

```

An "x" indicates that the data bit is xored into the corresponding CRC bit.
For example A is a xor of a,b,c,g,h,i and k.

The key is that there are at least two x's above each data bit, so a single data flip results in at least two CRC bits being changed and from that pattern one can deduce which data bit has been flipped.

The receiver after decoding the 15 bits at each FFT window would perform the error correction and feed the first 11 bits into the HDLC decoder. This approach gives us reduced data bandwidth (11 * 300 = 3300 bps) but makes a protection against single bit flips per one FFT window. Such flips would occur as a result of selective fading, for example, when one of the carriers gets much weaker. Still we don't have protection against pulse-type noise, which would wipe out most likely several bits in one FFT window.

Encoding approach #2 (Protection against error bursts)
=====

Interleave the bits ! Well... interleaving requires the receiver to synchronize to the interleave frames... or maybe not ?
We "interleave" the bit the following way:

```

      1 2 3 .... <- window number
1   a l w . . . . .
2   . b m x . . . .
3   . . c n y . . .
4   . . . d o z . .
5   . . . . e p . .
6   . . . . . f q .
7   . .           . g r .
8   . .           . h s .
9   . .           . i t .
10  . .           . j u .
11  . .           . k v .
12  . .           . A E .
13  . . . . . . . . . B F .
14  . . . . . . . . . C G .
15  . . . . . . . . . D H .

^
carrier number

```

If say window #2 gets corrupted by a spike the flipped bits l and b will belong to different groups and the FEC will correct them in every group separately. Can this technique be actually called "interleaving" ?

For better protection one can make the "interleave" factor larger, like 4. Anyway, I believe it should be at least 2, as the bits decoding is differential (that is it relies on

the previous window). If a spike corrupts one FFT window this results in bit flips in that window and in the next one.

Summary
=====

I find the proposed scheme capable of correcting data corruption resulting from selective fading and pulse type noise.
The receiver needs just to synchronize to the bit clock
- it doesn't need to know about SYNC words, magic numbers, etc
(a higher level HDLC or another frame decoder needs to know that).
At same time the scheme is simple and not very demanding
in terms of CPU power or RAM storage.

The concrete numbers and the FEC method given here are just an example.
When we increase the number of carriers and make the FEC "stronger"
(soft decisions ?) the coding would become certainly more efficient.
And of course we can run Reed-Salomon on top of that layer...

Did I reinvent the wheel ? Sure, I did :-)
Comments ?

Pawel

9. A coding idea for a parallel carrier modem

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 15 Feb 1995 14:50 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <6C0320C5606000E2@chopin.ifj.edu.pl>
Rendered body type: text/plain

Just to stimulate the discussion I would like to say an idea I got recently about encoding bits and frames onto a multi carrier signals.

As a base I take the approach to have several regularly spaced carriers, each being modulated in phase.
To decode/encode these signals we do "sliding window" FFT.
Each window overlaps the one before and the one after by half of its size.
At every window we change the carrier's phase by +90 or -90 degrees to encode 0s or 1s respectively. We are limited to 90 degrees shifts to keep adjacent symbols fully independent - this is because the windows overlap. Thus we encode one bit per FFT window.

Just to make things more concrete let's assume we take 64 samples per window at 9600 Hz sampling rate. This gives up 32 carriers spaced by 150 Hz.
We only use 15 carriers: the first at 600 Hz and the last at 2700 Hz to fill up the 300-3000 Hz spectrum.
There are 300 FFT windows/second which gives up 300 bits/second/carrier.
Total raw bit rate would be $300 * 15 = 4500$ bps.

Encoding approach #0
=====

Let's take the bits from say the HDLC encoder:

abcdefghijklmnopqrstuvwxy....

and code them straight onto the 15 carriers - like this:

```
1 2 3 .... <- window number
1 a p .
2 b q .
3 c r .
4 d s .
5 e t .
6 f u .
7 g v .
8 h w .
9 i x .
10 j y .
11 k z .
12 l . .
13 m . .
14 n . .
15 o . .
```

^

carrier number

The receiver (after achieving the bit synchronization) would take the FFT of every window, compare the phases with the previous window, decode 15 bits and feed them (with a predetermined order) into the HDLC decoder. This approach gives us the full data bandwidth available but no protection against bit flips made by noise.

Encoding approach #1 (We make up a simple FEC)

=====

We only send 11 data bits per FFT window and the remaining 4 bits we use for a simple FEC.

```
1 2 3 .... <- window number
1 a l w
2 b m x
3 c n y
4 d o z
5 e p .
6 f q .
7 g r .
8 h s .
9 i t .
10 j u .
11 k v .
12 A E .
13 B F .
14 C G .
15 D H .
```

^
carrier number

ABCDEFGH... are the CRC bits designed to correct single bit flips.

My (possibly naive) way to compute the CRC bits would be:

```
x x x      x x x  x  A
x      x x  x x  x x  B
      x  x  x x  x x x  C
        x  x x  x x x x  D
                                ^ CRC bits
a b c d e f g h i j k <- data bits
```

An "x" indicates that the data bit is xored into the corresponding CRC bit. For example A is a xor of a,b,c,g,h,i and k.

The key is that there are at least two x's above each data bit, so a single data flip results in at least two CRC bits being changed and from that pattern one can deduce which data bit has been flipped.

The receiver after decoding the 15 bits at each FFT window would perform the error correction and feed the first 11 bits into the HDLC decoder. This approach gives us reduced data bandwidth (11 * 300 = 3300 bps) but makes a protection against single bit flips per one FFT window. Such flips would occur as a result of selective fading, for example, when one of the carriers gets much weaker. Still we don't have protection against pulse-type noise, which would wipe out most likely several bits in one FFT window.

Encoding approach #2 (Protection against error bursts)

=====

Interleave the bits ! Well... interleaving requires the receiver to synchronize to the interleave frames... or maybe not ? We "interleave" the bit the following way:

```
1 2 3 .... <- window number
1 a l w . . . . .
2 . b m x . . . .
3 . . c n y . . .
4 . . . d o z . .
5 . . . . e p . .
6 . . . . . f q .
7 . . . . . g r .
8 . . . . . h s .
```

```

9      . . . . . i t .
10     . . . . . j u .
11     . . . . . k v .
12     . . . . . A E .
13     . . . . . B F .
14     . . . . . C G .
15     . . . . . D H .

```

^
carrier number

If say window #2 gets corrupted by a spike the flipped bits l and b will belong to different groups and the FEC will correct them in every group separately. Can this technique be actually called "interleaving" ?

For better protection one can make the "interleave" factor larger, like 4. Anyway, I believe it should be at least 2, as the bits decoding is differential (that is it relies on the previous window). If a spike corrupts one FFT window this results in bit flips in that window and in the next one.

Summary
=====

I find the proposed scheme capable of correcting data corruption resulting from selective fading and pulse type noise. The receiver needs just to synchronize to the bit clock - it doesn't need to know about SYNC words, magic numbers, etc (a higher level HDLC or another frame decoder needs to know that). At same time the scheme is simple and not very demanding in terms of CPU power or RAM storage.

The concrete numbers and the FEC method given here are just an example. When we increase the number of carriers and make the FEC "stronger" (soft decisions ?) the coding would become certainly more efficient. And of course we can run Reed-Salomon on top of that layer...

Did I reinvent the wheel ? Sure, I did :-)
Comments ?

Pawel

10. Re: [HFSIG:243] More Piccolo info from G4ZHZ

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 15 Feb 1995 09:33:05 PST8PDT
To: hfsig@tapr.org
Message-ID: <20AAA631BC1@frl.orst.edu>
Rendered body type: text/plain

Hi Frode/Adrian,

I am very pleased to hear about this outstanding work. Adrian must have devoted a great deal of his time to this project. I'll be happy to assist where I can. Hope he manages to subscribe to the SIG.

```

> I am posting yet another message from Adrian Nash, G4ZHZ.
> I hope to get him to join the list in the next few days.
>
>
> Hello Again
>
> Thanks for your email. Yes, I am interested in a copy of the MIL-STD-188
spec.
> I would be very grateful if you could send me a photocopy. I will look up
for
> the Cardinal sound card and see if I can get one and adapt my software.
> When I have done this, I will release the source code for the Cardinal
Sound
> card. The problem is that I suspect that the 2115 is an ADSP-2105 with a
host
> port added. This means that it will have 512 words of data RAM and 1k of
> program RAM. It may be necessary to "prune" the code to fit in this size.
>

```

<..... lines deleted.....>

Yes, that is correct about the 2115. The sound card has some 15KW (24-bit) zero wait external program memory and 8KW (16-bit) zero wait data memory that you will find will meet your needs.

Perhaps if Adrian could E-mail me directly, we can take up porting issues and hardware needs.

73's

Johan, KC7WW

11. Re: [HFSIG:244] Coding idea for a parallel carrier modem

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 15 Feb 1995 09:36:07 PST8PDT
To: hfsig@tapr.org
Message-ID: <20AB7C14839@frl.orst.edu>
Rendered body type: text/plain

Dear Pawel,

Welcome! Your contribution to the SIG is most welcome too. I am delighted to see all the good ideas recently.

A few comments, for what its worth:

Pawel Jalocha wrote:

> Just to stimulate the discussion I would like to say an idea I got recently
> about encoding bits and frames onto a multi carrier signals.
>
> As a base I take the approach to have several regularly spaced
> carriers, each being modulated in phase.
> To decode/encode these signals we do "sliding window" FFT.
> Each window overlaps the one before and the one after by half of its size.
> At every window we change the carrier's phase by +90 or -90 degrees
> to encode 0s or 1s respectively. We are limited to 90 degrees shifts
> to keep adjacent symbols fully independent - this is because the windows
> overlap. Thus we encode one bit per FFT window.
>

< some lines deleted >

I think we are basically talking about one variant of orthogonal frequency division multiplexing (OFDM)? Hopefully I can interest you in building on the example modem that I have put on the hfsig upload area? This is very much along the ideas that you have described and I am sure you will have a lot of fun with it. I'll make the source code available if you are interested - this will give you a good starting place?

Although the ideas on coding are quite reasonable and feasible, I am of the opinion that we should put our heads together and implement one of the modern schemes that were discussed previously on this forum. Perhaps you would like to review these ideas and give us some further feedback?

> Did I reinvent the wheel ? Sure, I did :-)
> Comments ?

> Pawel

Hope to hear from you soon,

73's
Johan, KC7WW

12. (no subject)

From: Edson W Pereira <ewp@world.std.com>
Date: Wed, 15 Feb 1995 19:16:45 +0001 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.3.89.9502151917.A26747-0100000@world.std.com>
Rendered body type: text/plain

subrcrbe rfsig

13. Re: [HFSIG:247] Re: Coding idea for a parallel carrier modem

From: JALOCHA@chopin.ifj.edu.pl
Date: Thu, 16 Feb 1995 16:57 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <46E7306640600F04@chopin.ifj.edu.pl>
Rendered body type: text/plain

>I think we are basically talking about one variant of orthogonal frequency
>division multiplexing (OFDM)?

Yes, that's it.

To answer this question I needed to look quite some messages backwards
to decode what the "OFDM" term means exactly :-)

>Hopefully I can interest you in building
>on the example modem that I have put on the hfsig upload area? This is
>very much along the ideas that you have described and I am sure you will
>have a lot of fun with it. I'll make the source code available if you are
>interested - this will give you a good starting place?

For the moment the only DSP platform I can work on is the DSPCARD4.
I have some start pieces like the FFT code working together with sliding
window (a left-over from noise suppressors).
I only need to adapt the "sliding window" so the receiver part is allowed
to shift in time for symbol-level synchronization.

>Although the ideas on coding are quite reasonable and feasible, I am of
>the opinion that we should put our heads together and implement one of
>the modern schemes that were discussed previously on this forum. Perhaps
>you would like to review these ideas and give us some further feedback?

What's "not modern" in my ideas ? :-)

I just got one idea of a very simple coding for real weak signals.
Consider my previous example of 15 carriers each carrying 300 bps.
If we just transmit same stream of bits on every carrier, interleaved
in time the way I described. The receiver would do majority vote
(or better sum up all the bits at the analog level and then do
the hard decision). This way we pass 300 bps in a voice channel
in a "cruel" way but with high level of redundancy.
Unfortunately I am not able to make a theoretical comparison
to some other schemes. Could somebody please do the calculations ?

Pawel

14. Re: [HFSIG:242] Re: HFSig Status

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Thu, 16 Feb 95 13:23:15 nst
To: hfsig@tapr.org
Message-ID: <9501167929.AA792969812@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

Johan Wrote:

>Personnaly, I have been experimenting with the OFDM idea, i.e.
>parallel-tone approach. My program uses 8/8 channels, each channel
>being QDPSK, symbol rate being 31.24 Hz (2*8*31.25=500 bps raw) in a
>250 Hz channel. It uses a FFT-based demodulator. The idea is, DSP
>resources permitting, to expand to 32 channels (1000 bps raw rate). A
>rate 1/2 ECC will be added. In addition some form of ARQ data-link
>layer is being considered.
>
>If you are interested - a demo version for this modem running on the
>sound card is available from the TAPR file server: ftp.tapr.org
>tapr/SIG/hfsig/upload/parmdm1.zip

Johan,

I retrieved the file, but it is in binary .exe form. Would it be
possible to obtain the source form for evaluation?

ftp.tapr.org tapr/SIG/hfsig/upload/parmdm1.zip (pmodem1.zip)

Also, it seems that most of you (??) are using the same sound card for
developments. Please give its spec. and I'll order one. Alot easier to
share experiments. Possibly some long distance trials?

I operate in the 2-30MHz range on Marine HF Radios (Mostly 3-12MHz), with both a whip and a folded L antenna on the building roof here in town. Fancier tests may be made from our radio station down the coast (~1hr drive). I usually use Voice 300-2700Hz Bands.

regards (is this what 73's means??)

>-----
> Paul G Russell
> St. John's, Newfoundland, Canada
> paulr@neweast.ca
> Fax: Canada (709)576-2125
>
> "Engineering is a Profession in laziness - How to get
> the most done with the least amount of effort" - PGR
>
> "You have got to be half crazy, otherwise all the nuts
> in the world will drive you totally insane" - PGR
> corollary: "Which one am I?"
>
> "Oops, Shouldn't have done that..." - PGR et al
>
> Oh yea, my opinions are mine, and maybe others as well?
>-----

15. Re: [HFSIG:249] Re: Coding idea for a parallel carrier mode

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 16 Feb 1995 09:05:56 PST8PDT
To: hfsig@tapr.org
Message-ID: <22237720259@frl.orst.edu>
Rendered body type: text/plain

Dear Pawel,

<.... some lines deleted >

The past message thread on ODFM would be interesting to you, however, you are indeed on the right track.

>
> For the moment the only DSP platform I can work on is the DSPCARD4.
> I have some start pieces like the FFT code working together with sliding
> window (a left-over from noise suppressors).
> I only need to adapt the "sliding window" so the receiver part is allowed
> to shift in time for symbol-level synchronization.
>

I am also looking at the new M56002 EVM, but too early to say anything yet.

Be sure to share your ideas on FFT-based noise reduction. I, and others, have played with LMS and Wiener filters that appear to do a good job on HF, however, have been planning to work on the FFT-based method. These ideas come in handy for HF - this is a good place to discuss them too.

> What's "not modern" in my ideas ? :-)

Oh no - you are a renaissance man, Pawel. What I mean is that we should put our joint efforts towards those schemes that yields the highest returns. Simple CRC's, for example is only the first (weak) line of defence. We have to include both error detection and correction. That probably means convolutional coding or block ECC codes with extensive interleaving as you correctly indicated. Soft-decision decoding is probably also a good idea. However, I am just repeating what you can read in the past message threads for the SIG. I hope I am forgiven?

>
> I just got one idea of a very simple coding for real weak signals.
> Consider my previous example of 15 carriers each carrying 300 bps.
> If we just transmit same stream of bits on every carrier, interleaved
> in time the way I described. The receiver would do majority vote
> (or better sum up all the bits at the analog level and then do
> the hard decision). This way we pass 300 bps in a voice channel
> in a "cruel" way but with high level of redundancy.
> Unfortunately I am not able to make a theoretical comparison
> to some other schemes. Could somebody please do the calculations ?

Just for interest sake: the 39-tone MIL-STD-188 modem implements some

of this. It uses 39 tones, spread out over 3kHz, each tone is phase modulated. At the highest rate, each channel represents a single bit stream - when needed, the bit rate is dropped in favour of redundancy, where bit streams are duplicated, i.e., appears in several channels. This then allows for both frequency and time diversity - very effective. My experiences with the analog summing idea, such as implemented in Pactor memory-ARQ, is that it is very weak, perhaps 1/2 dB at best in a non-AWGN channel.

Keep up the good work.

73's

Johan, KC7WW

16. TAPR.ORG Downtime

From: Greg Jones <gjones@tenet.edu>
Date: Thu, 16 Feb 1995 11:25:44 -0600 (CST)
To: tapr-bb@tapr.org (TAPR-BB mailing), hfsig@tapr.org (HF SIG mailing),
netsig@tapr.org (NETSIG mailing), bbssig@tapr.org (BBS SIG mailing),
fccreg@tapr.org (fcc), tapr-board@tapr.org (TAPR Board),
pcs@tapr.org (TAPR PCS Project), dsp-93@tapr.org (DSP-93 Build),
dsp93dev@tapr.org (DSP93 Development), aprssig@Alice-Thurman.tenet.edu,
tuc52@tapr.org (TUC-52), tnc95@tapr.org (TNC-95)
Message-ID: <199502161725.LAA03238@Alice-Thurman.tenet.edu>
Rendered body type: text/plain

TAPR.ORG Downtime 2/16/95

The TAPR.ORG system will be moving locations on Monday morning, February 20th, 1995.

During this move, the system will be unavailable. A second message will be posted when the system is fully operational.

17. Re: [HFSIG:250] Re: HFSig Status

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 16 Feb 1995 10:05:24 PST8PDT
To: hfsig@tapr.org
Message-ID: <22335256DDD@frl.orst.edu>
Rendered body type: text/plain

Hi Paul,

Pardon the amateur radio terminology, I did not realize that you were not a licenced radio amateur - how about getting a licence for future on-the-air tests?

<... some lines deleted ...>

> I retrieved the file, but it is in binary .exe form. Would it be
> possible to obtain the source form for evaluation?
>
> ftp.tapr.org tapr/SIG/hfsig/upload/parmdm1.zip (pmodem1.zip)
>

Since several asked for it, I placed the source code in PMODEM2.ZIP.

>
> Also, it seems that most of you (??) are using the same sound card for
> developments. Please give its spec. and I'll order one. Alot easier to
> share experiments. Possibly some long distance trials?
>

You have to look for a sound card with the PSA chipset. These include the Cardinal Pro 16, Orchid Soundwave32, Western Digital Paradise 16-DSP, Adaptec AMExxxx, Wearness Beethoven, and Echo Speech. I recently ordered some Cardinal Pro 16 cards from J&R Music (800-221-8180) for \$39.95 on a closeout sale. Not sure how long that will last. Give it a try if you wish.

These cards are basically a 16 MIPS ADSP-2115, memory, and stereo CODEC. There also are other alternatives such as the Connection+ phone modems from Digicom that has the same DSP chip, though a different CODEC. There are some development software around for those as well. I have not,

however, looked into those at all.

Hope this helps.

--Johan

18. Re: [HFSIG:253] Re: HFSig Status

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Thu, 16 Feb 95 15:20:14 nst
To: hfsig@tapr.org
Message-ID: <9501167929.AA792976911@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

Johan Wrote:

><... some lines deleted ...>

>Since several asked for it, I placed the source code in PMODEM2.ZIP.

Oops somewhere - The .zip is actually the text description file,
please upload it again.

>Pardon the amateur radio terminology, I did not realize that you
>were not a licenced radio amateur - how about getting a licence for
>future on-the-air tests?

I hadn't thought about it .. maybe.. you mean learn more new stuff??!!
I'll download some of the requirements files to find out just what is
involved.
And since this is a ham List, I should use your terminology (meters vs
MHZ... etc.)
As for Testing, the Company has several assigned frequencies between
3&20MHz (100m -- 15m ??) that we could use.

Those Cardinal Cards are all sold out. They have the Orchid
Soundwave32 for \$159.00 This is fully code compatible I assume??
I'll order one tomorrow.

-Paul Russell

19. modem for E-M-E?

From: jmorriso@bogomips.ee.ubc.ca (John Paul Morrison)
Date: Thu, 16 Feb 1995 10:17:10 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0rfAlL-0004gQC@bogomips.ee.ubc.ca>
Rendered body type: text/plain

Would the modulation techniques being discussed here work with EME (moonbounce)?

btw the X-Comment: for the list is wrong :-)
Tucson Amateur Packet Radio HF Speical Interest Group
^^^^^^

BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --
jmorriso@bogomips.ee.ubc.ca ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org

20. Replacement PMODEM3.ZIP

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 16 Feb 1995 11:16:49 PST8PDT
To: hfsig@tapr.org
Message-ID: <22466256F9D@frl.orst.edu>
Rendered body type: text/plain

Hi Paul,

OOPS - sorry about that. Try pmodem3.zip. Sizes look OK now.

Johan, KC7WW

21. Re: [HFSIG:255] modem for E-M-E?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 16 Feb 1995 11:34:24 PST8PDT
To: hfsig@tapr.org

Message-ID: <224B0C32601@frl.orst.edu>

Rendered body type: text/plain

Hi John,

John Paul Morrison wrote:

> Would the modulation techniques being discussed here work
with EME (moonbounce)?

Phil mentioned in one of his early postings on multi-tone schemes,
that his suggestion originated from EME ideas, so guess there may
be some usable ideas here.

For HF, we are basically talking about low symbol rates combined
with time and frequency interleaved methods to combat the nature of
the HF channel. I am no expert at EME, but from what I have read, you are
dealing with very unfavorable S/N (closer to an AWGN channel than HF),
as well as deep fading phenomena. Although our methods would probably
be usable, probably not quite addressing the same problems. You may need
to use similar modulation combined with convolutional coding with very
large constraint factor - just my \$0.02.

Any other comments?

> btw the X-Comment: for the list is wrong :-)
> Tucson Amateur Packet Radio HF Speical Interest Group
^^^^^^

Could someone tell us how to fix this? Thanks.

73's

Johan, KC7WW

22. Re: [HFSIG:257] Re: modem for E-M-E?

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)

Date: Thu, 16 Feb 95 14:19:19 CST

To: hfsig@tapr.org

Message-ID: <9502162019.AA02563@eagle.aud.alcatel.com>

Rendered body type: text/plain

> Hi John,

>

>

> John Paul Morrison wrote:

>

> > Would the modulation techniques being discussed here work
> with EME (moonbounce)?

>

> Phil mentioned in one of his early postings on multi-tone schemes,
> that his suggestion originated from EME ideas, so guess there may
> be some usable ideas here.

>

> For HF, we are basically talking about low symbol rates combined
> with time and frequency interleaved methods to combat the nature of
> the HF channel. I am no expert at EME, but from what I have read, you are
> dealing with very unfavorable S/N (closer to an AWGN channel than HF),
> as well as deep fading phenomena. Although our methods would probably
> be usable, probably not quite addressing the same problems. You may need
> to use similar modulation combined with convolutional coding with very
> large constraint factor - just my \$0.02.

>

> Any other comments?

>

> Johan, KC7WW

>

If we are talking strictly AWGN, then it seems there might be two choices:

- 1) If bandwidth is no problem, then m-ary FSK provides the best
possible BER vs. SNR for large values of m, approaching
Shannon's limit of 'arbitrarily low error rate' at
-1.6 dB Eb/No for m=infinite. For finite values of
large m, convolutional coding and soft-decisions should
improve things, but I don't know how much.

- 2) If bandwidth is of some concern, then coherent 2PSK, with
convolutional coding, and soft-decision is optimal within

a bandwidth no wider than the Nyquist limit
(channel width of $1/T$ for $\alpha=0$). Coding is almost
always a good investment of bandwidth in AWGN.

Perhaps there are more recent developments that make these two
limiting cases invalid -- any information on such would be of interest!

Is there another characteristic of the E-M-E path than makes AWGN
a poor assumption?

- Tom, N5EG

23. Re: [HFSIG:257] Re: modem for E-M-E?

From: Greg Jones <gjones@tenet.edu>
Date: Thu, 16 Feb 1995 15:26:44 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199502162126.PAA04505@Alice-Thurman.tenet.edu>
Rendered body type: text/plain

According to Johan Forrer:
> > btw the X-Comment: for the list is wrong :-)
> > Tucson Amateur Packet Radio HF Speical Interest Group
> ^^^^^^^
> Could someone tell us how to fix this? Thanks.

This is one for me -- I'll get the correction submitted.

Greg

24. RE: [HFSIG:258] Re: modem for E-M-E?

From: Stephen McNeill <srghsjm@GIH.GRACE.CRI.NZ>
Date: Fri, 17 Feb 1995 09:36:13 EST
To: hfsig@tapr.org
Cc: srghsjm@GIH.GRACE.CRI.NZ
Message-ID: <0098C1D2.5EF49580.3415@GIH.GRACE.CRI.NZ>
Rendered body type: text/plain

In an earlier message Johann said:

> For HF, we are basically talking about low symbol rates combined
> with time and frequency interleaved methods to combat the nature of
> the HF channel. I am no expert at EME, but from what I have read, you are
> dealing with very unfavorable S/N (closer to an AWGN channel than HF),
> as well as deep fading phenomena. Although our methods would probably
> be usable, probably not quite addressing the same problems. You may need
> to use similar modulation combined with convolutional coding with very
> large constraint factor - just my \$0.02.

Er, I'm no expert on EME either. I've followed most of what's been on
this group, but what is AWGN? Did it come up before and did I miss it?

EME would-be data users tend to be in two camps: those who would like to
be able to achieve as high a bit rate as possible for a given power, and
those who would like to achieve communication at as low a power as possible,
at ridiculously low bit rates.

Both are formidable challenges, but I'm interested in the latter.

Stephen ZL4HG

25. Re: [HFSIG:251] Re: Coding idea for a parallel carrier mode

From: JALOCHA@chopin.ifj.edu.pl
Date: Fri, 17 Feb 1995 12:33 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <EB2E2E95E0601FE0@chopin.ifj.edu.pl>
Rendered body type: text/plain

> Be sure to share your ideas on FFT-based noise reduction. I, and others,
> have played with LMS and Wiener filters that appear to do a good job on HF,
> however, have been planning to work on the FFT-based method. These ideas
> come in handy for HF - this is a good place to discuss them too.

Well, I made two noise filters:

One is based on the autocorrelation function. The autocorrelation
is computed in a continuous fashion, then it is used to make a FIR
filter and the audio is passed through this filter.
For example for a continuous sine wave, the autocorrelation is a cosine

of same frequency (adding noise makes minimal changes).
If you put that cosine into the FIR coefficients (you apply a window on the way) your filter will pass only that frequency.
I find this type of filter not good for speech, but for CW it might be just what you need. Unfortunately I never got any comments...

The other filter is based on the FFT. A 512 point FFT is being continuously (by sliding window) computed on the input signal. Then some more or less sophisticated criteria (numerous user-selectable options) are applied to every frequency bin in order to decide whether it should be muted or not. This filter works well for speech (even music with properly selected parameters). If somebody wishes to talk about the details I am ready...

I have a code for both filters running on my DSPCARD4.
I will attempt a port to the 56002 evaluation board, after I get one.

>Oh no - you are a renaissance man, Pawel. What I mean is that we should
>put our joint efforts towards those schemes that yields the highest
>returns. Simple CRC's, for example is only the first (weak) line of
>defence. We have to include both error detection and correction. That
>probably means convolutional coding or block ECC codes with extensive
>interleaving as you correctly indicated. Soft-decision decoding is probably
>also a good idea. However, I am just repeating what you can read in the past
>message threads for the SIG. I hope I am forgiven?

My ideas are simple but do provide forward error correction with time and frequency diversity. I don't want to overcome those great ideas - my intention is to provide simple coding schemes for initial tests and for future reference.
After all, it's difficult to jump straight onto those sophisticated methods. I imagine, we develop an OFDM modem in few steps:
1. No FEC - just split the data bits among the carriers.
2. "Simple" FEC with time and frequency diversity.
3. "Sophisticated" coding to get the last dB's out of the noise.

When you have these, you let an average amateur compare what are the gains and possibly drawbacks :-) of various schemes.
And you might make them laugh when they find that scheme #1 works best :-)

>My experiences
>with the analog summing idea, such as implemented in Pactor memory-ARQ, is
>that it is very weak, perhaps 1/2 dB at best in a non-AWGN channel.

Analog summing is indeed "risky" on nowadays HF: if you get a strong continuous carrier in your passband it will overcome all the other "analog" bits (but "smart" techniques could get around this).
This is not a problem in deep space communication I guess : there, hearing CW is like finding a Chevrolet on the Moon :-) (or an ETI...).

OK. Enough talking... I have a _real_ problem now. I was coding yesterday the FFT part of the OFDM modem and got almost stuck at one point:
which window shape shall I use for "sliding window" ?
If I use a nice and smooth window I get strong inter-carrier crosstalks.
I suspect the rectangular window (that is no window at all) would make minimal crosstalk. Could somebody help ?

Pawel

26. Re: [HFSIG:261] Re: Coding idea for a parallel carrier mode

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 20 Feb 1995 13:02:54 PST8PDT
To: hfsig@tapr.org
Message-ID: <2862D444D4A@frl.orst.edu>
Rendered body type: text/plain

Hi Pawel,

Some feedback:

>
>One is based on the autocorrelation function. The autocorrelation
>is computed in a continuous fashion, then it is used to make a FIR
>filter and the audio is passed through this filter.
>For example for a continuous sine wave, the autocorrelation is a cosine
>of same frequency (adding noise makes minimal changes).
>If you put that cosine into the FIR coefficients (you apply a window
>on the way) your filter will pass only that frequency.
>I find this type of filter not good for speech, but for CW it might
>be just what you need. Unfortunately I never got any comments...

>

The LMS (least mean squares) algorithm works very well on voice. If you are interested in a really interesting expose on the finer points of doing quality filtering - obtain W9GR's code that goes with his QEX/QST articles. The articles does not tell you the tricks, however, reading the code is very educational.

>The other filter is based on the FFT. A 512 point FFT is being continusly
> (by sliding window) computed on the input signal. Then some more or less
>sophisticated criteria (numerous user-selectable options) are applied
>to every frequency bin in order to decide whether it should be muted or
not.
> This filter works well for speech (even music with properly selected
>parameters). If somebody wishes to talk about the details I am ready...
>

I am queruious about the sample rate are you using Pawel. I thought about doing something like this, though found that I, either had to work at a lower sample rate, or work with overlapping FFT windows and use up-sampling at the output stage (i.e. the output rate of the FFT processor is lower than the input sample rate, thus the need for an interpolator filter to output the results of the IFFT.)

> OK. Enough talking... I have a _real_ problem now. I was coding yesterday
>the FFT part of the OFDM modem and got almost stuck at one point:
>which window shape shall I use for "sliding window" ?
> If I use a nice and smooth window I get strong inter-carrier crosstalks.
>I suspect the rectangular window (that is no window at all) would
>make minimal crosstalk. Could somebody help ?
>

Good question. The basic reason is the requirement for nyquist signaling are changed when you convolve with a window function other than a rectangular window. Do not use anything but the rectangular window for this type of demodulator.

73's

Johan, KC7WW

27. TAPR.ORG Location Change

From: Greg Jones <gjones@tenet.edu>
Date: Mon, 20 Feb 1995 15:23:15 -0600 (CST)
To: tapr-bb@tapr.org (TAPR-BB mailing), bbssig@tapr.org (BBS SIG mailing),
netsig@tapr.org (NETSIG mailing), dsp-93@tapr.org (DSP-93 Build),
hfsig@tapr.org (HF SIG mailing), aprssig@tapr.org
Message-ID: <199502202123.PAA09624@Joyce-Perkins.tenet.edu>
Rendered body type: text/plain

TAPR.ORG has been relocated and has a new numeric IP address of:

204.96.214.85

DNS servers should have been updated starting yesterday....but just in case yours is a little slow, you can use the numeric IP until things get fully updated.

The new location allows for a faster connection into Internet which should help the site in the long run.

Thanks to Lee, NSLYT, for his continued help and effort on the TAPR.ORG site.

Cheers - Greg

28. HF channel simulator - DSP code

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 20 Feb 1995 14:10:53 PST8PDT
To: HFSIG@tapr.org
Message-ID: <2874F2E1E69@frl.orst.edu>
Rendered body type: text/plain

Hi All,

I have uploaded a very experimental piece of DSP code for the

HF channel simulator. This is basically for those wishing to participate in developing the HF channel simulator.

You would need a PC sound card (PSA chipset) and the freeware spasm21 assembler for doing further work on the DSP code. A C compiler is required to develop the PC code.

Good luck!

Johan, KC7WW

29. Re: [HFSIG:244] Coding idea for a parallel carrier modem

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 21 Feb 1995 02:59:26 -0800
To: JALOCHA@chopin.ifj.edu.pl
Cc: hfsig@tapr.org
Message-ID: <199502211059.CAA08066@unix.ka9q.ampr.org>
Rendered body type: text/plain

Pawel,

A few (hopefully helpful) comments on your proposal.

Your baud rate and bits/hz figures seem rather optimistic for all but the best HF channels. (See previous discussions about ionospheric phase distortion and bandwidth "efficiency" vs power performance).

Your FEC code appears to be a (15,11) Hamming code, one of the first block FEC codes invented. All Hamming codes have a minimum distance between codewords of 3 (any code word differs from all others in at least 3 places) and so can correct one error per codeword.

Hamming codes are not particularly strong. Much better (i.e., more error correcting capability and/or lower overhead) codes have since been invented.

Part of the problem is the small block size of your code. All other things being equal, bigger blocks (if you can decode them) perform better. This has been known since Shannon, so the trick has always been to discover large block codes that are practical to generate and decode. The Reed-Solomon codes are probably the best known result.

Your code has a rate of 11/15 or .733. So let's examine a Reed-Solomon code with the same rate. I'll pick the (255,187) code over GF(256). (RS codes over GF(256) are popular since they give you a convenient 8-bit symbol size. This in turn implies the 256-1=255 byte blocksize.)

This RS code has 255-187 = 68 parity symbols. Any Reed-Solomon code can correct up to one-half as many errors per block as it has parity symbols. So for this code it can correct up to 34 symbols. That's 34/255 = 13.3% of the encoded block. Your code can correct only 1/15 = 6.7% of the block.

Also, the RS code doesn't mind if all those errors occur in one big burst of 34x8 = 272 bits, without any interleaving at all. In fact, it *prefers* errors to occur in bursts because it doesn't matter how many errors occur in a single Reed-Solomon symbol - even one bad bit destroys the symbol, so you might as well heap your other errors there too. This explains the well-known burst error correcting capability of Reed-Solomon codes. (Interleaving is still often desirable with RS coding, though, to reduce the possibility of a single long burst beyond this code's capability.)

But if we consider the pathological case of the bit errors each being in a separate symbol, this code could correct only 34 bit errors per block, a bit error rate of only 1.7%. So you really have to work out the probabilities knowing something about your channel and its error statistics before you can decide on a good code.

Phil

30. Re: [HFSIG:262] Re: Coding idea for a parallel carrier mode

From: JALOCHA@chopin.ifj.edu.pl
Date: Tue, 21 Feb 1995 12:00 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <0B32A7F580603E82@chopin.ifj.edu.pl>
Rendered body type: text/plain

>I am quervious about the sample rate are you using Pawel. I thought about
>doing something like this, though found that I, either had to work at a
>lower sample rate, or work with overlapping FFT windows and use up-sampling
>at the output stage (i.e. the output rate of the FFT processor is lower
>than the input sample rate, thus the need for an interpolator filter
>to output the results of the IFFT.)

OK. Here are some details. Let's assume for the moment that
I use 9600 kHz sampling and a 512 point FFT.
I used sliding window of 512 samples in steps of 256 samples.

The steps are:

1. take 512 samples starting at sample #0
2. multiply these by a sine window (a sine wave from 0 to pi)
3. do the FFT (512 real samples -> 256 complex frequency-domain points)
4. process the FFT (remove "useless" frequency components)
5. do the Inverse FFT (we get back 512 real samples)
6. multiply again by the sine window
7. place these 512 samples at the outbut buffer starting at #0

The next cycle is:

1. take 512 samples starting at sample #256 (note the index moved by 256 not 512))
...steps 2-6 are identical...
7. add these 512 samples to the output buffer starting at #256
(note we add samples to what is already in the output buffer).

And so on... the next cycles would start at 512, 768, 1024, etc.

Note that if you skip points 3, 4 and 5 this algorithm effectively
doesn't do any change to the input stream (well, it does change the first
256 samples but afterwards there is really no change).
This is because I use sine-shaped window and I apply it twice during
the process so the effective window for every batch (512 samples)
is $\sin(x)^2$. Now imagine several $\sin(x)^2$ shifted by 90 degrees
- they form a flat line.
Not a clear explanation, but I hope you can dig out the merits :-)

Now we put in points 3 and 5 and still we should see no change
in the data stream as FFT+IFFT doesn't change the batch contents.

>Good question. The basic reason is the requirement for nyquist signaling are
>changed when you convolve with a window function other than a rectangular
>window. Do not use anything but the rectangular window for this type
>of demodulator.

I agree that with rectangular window we get no crosstalk.
However, when you get miss-tuned the crosstalk would appear
and not only in the adjacent carriers.
Strange things would happen too when you are "miss-tuned" in the symbol
timing...

Looking from another point: if we send out data using the reactangular window
our symbols will have a large spread in frequency spectrum.
My intuition tells me, the symbol will have half of it's energy
out of the band intended for that symbol.
Remember that clicks you hear when listening to the demo multi-carrier modem ?

I tried to study this problem by numerical simulations using MathCAD
and I got to a conclusion that you should do proper shaping like $\sin(x)/x$
or similar. Then your symbols do occupy more-or-less the desired band
and the time and frequency cross-talk can be made reasonably low
and affecting only the adjacent carriers.
However the direct approach to proper shaping implies long FFTs...
that is for a 256 carriers modem you would need to make 1024
or 2048 point FFT. I hope there are smarter ways...

I'm sure there is somebody knoledgeable out there...

Pawel

31. Re: [HFSIG:249] Re: Coding idea for a parallel carrier modem

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 21 Feb 1995 03:09:54 -0800
To: JALOCHA@chopin.ifj.edu.pl
Cc: hfsig@tapr.org
Message-ID: <199502211109.DAA08095@unix.ka9q.ampr.org>
Rendered body type: text/plain

>I just got one idea of a very simple coding for real weak signals.
>Consider my previous example of 15 carriers each carrying 300 bps.
>If we just transmit same stream of bits on every carrier, interleaved
>in time the way I described. The receiver would do majority vote
>(or better sum up all the bits at the analog level and then do
>the hard decision). This way we pass 300 bps in a voice channel
>in a "cruel" way but with high level of redundancy.
>Unfortunately I am not able to make a theoretical comparison
>to some other schemes. Could somebody please do the calculations ?

This is basically "repetition coding". In a nonfading AWGN (additive white gaussian noise) channel this has no advantage over just sending the data once on a single carrier with full power. But it could have a real advantage on a channel with selective fading, assuming the transmitter and receiver are both linear and the receiver summing is done with sufficient precision such that faded bits are given low weight in the decision process.

You could do even better, though, with a real FEC code instead of simple repetition. It would be the same to the modem, but would provide considerably better power and bit error rate performance.

Phil

32. Re: [HFSIG:265] Re: Coding idea for a parallel carrier modem

From: JALOCHA@chopin.ifj.edu.pl
Date: Tue, 21 Feb 1995 12:38 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <10870C5260603E82@chopin.ifj.edu.pl>
Rendered body type: text/plain

Thanks for your remarks Phil,

>Your baud rate and bits/hz figures seem rather optimistic for all but
>the best HF channels.

I fully agree. My idea was: if the 300 bps packet works this scheme would work too providing some protection against selective fading, burst error and giving quite some bits/second. This is not along the idea to copy an unlistenable HF signal.

Going to more carriers makes the signaling rate lower (which is good) and that simple Hamming code carries less overhead. Just a little table:

carriers	data bits	CRC bits
15	11	4
31	26	5
63	57	6
127	120	7

Anyway that code can only correct single bits which is certainly not enough to copy if half of our band is wiped out by a speech signal...

Still, if I code a parallel carrier modem into my DSPCARD4 I will try this scheme first because of it's simplicity. Anyway, for the moment I just don't know enough details about RS (or any other) code to write the coder/encoder myself :-)

Pawel

33. Re: [HFSIG:258] Re: modem for E-M-E?

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 21 Feb 1995 03:46:25 -0800
To: hfsig@tapr.org
Message-ID: <199502211146.DAA08125@unix.ka9q.ampr.org>
Rendered body type: text/plain

> Is there another characteristic of the E-M-E path than makes AWGN
>a poor assumption?

Yeah, lots. Severe multipath fading, for one. The surface of the moon is extremely rough. Although the moon rotates on its axis exactly once per orbit of the earth, its orbit is slightly eccentric. So as the moon rotates at a constant rate on its axis, the small variation in its orbital velocity makes it seem to rock from side to side slightly as seen from earth. This "libration" causes the relative distances from the countless reflecting points on its surface to the user stations to not remain constant. A coherent CW carrier is typically

spread out 50 Hz or so at VHF/UHF frequencies by summing all those little random doppler variations.

Imagine a mirror ball from a dance hall, only much larger and thoroughly vandalized by smashing and dark grey spray painting. That's the moon at RF.

This effectively rules out any practical form of phase modulation and coherent detection. Noncoherent m-ary FSK would definitely seem to shine in this situation.

Phil

34. Re: [HFSIG:257] Re: modem for E-M-E?

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Tue, 21 Feb 1995 03:51:09 -0800
To: hfsig@tapr.org
Message-ID: <199502211151.DAA08128@unix.ka9q.ampr.org>
Rendered body type: text/plain
>dealing with very unfavorable S/N (closer to an AWGN channel than HF),

S/N does not define an AWGN channel. You can have AWGN channel with any S/N ratio. It simply means that the channel adds white gaussian noise, presumably at a constant power per unit bandwidth. Unless otherwise specified, the channel is otherwise completely benign - no fading, no interference, no phase distortion, nonlinearities, etc.

AWGN channels are the most easily analyzed when it comes to coding and modulation, but with the exception of satellite communications they are not, in general, good models for the real world.

Phil

35. Collection of ECC programs

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 21 Feb 1995 10:16:21 PST8PDT
To: hfsig@tapr.org
Message-ID: <29B67AA583E@frl.orst.edu>
Rendered body type: text/plain

Hi All,

I have placed a collection of error detection/correction programs from my library on ftp.tapr.org tapr/SIG/hfsig/download/ecc.zip.

These are all in C by various authors and includes Reed-Solomon, Golay, and BCH. These programs are a good place to start learning about this field.

Phil promised some convolutional code - how about it Phil?

Hope you find this of use.

73's

Johan Forrer, KC7WW

36. Digital downconverter IC

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Tue, 21 Feb 95 14:00:47 CST
To: hfsig@tapr.org
Message-ID: <9502212000.AA05061@eagle.aud.alcatel.com>
Rendered body type: text/plain

I received the 1994 Harris DSP Databook last week, and there is a very interesting DSP chip in it, the HSP50016. This part is an all-digital downconverter. It is capable of operating up to 52 MSPS. It contains a 32 bit NCO, with sin+cos outputs, capable of linear or chirp operation, and it multiplies that by the input words. Then it contains both I and Q decimation filters (with decimation range 64 to 131072), and I and Q FIR LPFs. The NCO, decimators and FIR are all programmable with a significant amount of resolution.

It appears the part would let one digitize the I.F. of a receiver, downconvert it to baseband, then decimate the output to a reasonable sampling rate. It has a bus interface to a microprocessor for programming. The input and output data is 16 bits, and the decimators go up to 66 bits

in width before scaling.

Seems like this might be an all-purpose digital receiver back end.
Call Harris Semiconductor Literature at 1-800-442-7747, or 407-724-3937
for more info or the DSP Databook. I've no idea what the cost is, comes
in 48 PGA or 44 PLCC.

#include <disclaimer.h>.

- Tom, N5EG

37. Re: [HFSIG:266] Re: Coding idea for a parallel carrier mode

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 21 Feb 1995 17:07:52 PST8PDT
To: hfsig@tapr.org
Message-ID: <2A2434209A5@frl.orst.edu>
Rendered body type: text/plain

Hi Pawel,

Good work!

> OK. Here are some details. Let's assume for the moment that
> I use 9600 KHz sampling and a 512 point FFT.
> I used sliding window of 512 samples in steps of 256 samples.

Thanks for the explanation - I got it now.
Another question: could you define "useless" frequency components?

OFDM: You are right about the problems during phasing. May help to look
at the MIL-STD-188 documentation for ideas. They go into some
detail on what a sync sequence look like. For one, they do not use
adjacent channel tones during sync time.

I also played around a bit with a simulation program to investigate the
OFDM waveform using raised cosines - the results were not encouraging,
like you indicated. You trade off performance (assuming you have sync)
for the ability to sync/bit-track on-the-fly. I am not sure you want
that when dealing with weak signals. There will be penalties, however,
especially under multipath conditions.

You may also get some ideas from J.D. Ralphs' book on MFSK signaling.
In one example he suggests a sync sequence to obtain bit sync, after which
the sync detector stays locked for the remainder of the frame. By
definition, this is non-coherent detection, however, in this situation, bit
sync also implies phase sync, and thus very close to matched filtering in
reality. Adrian knows the Piccolo stuff - perhaps he cares to explain how
he understands this issue.

The clicks in the OFDM demo is definately due to summing of 39 wave-
forms from a zero initial phase - the worst you can get. By just changing
the message content, one can actually hear the difference. I am fairly
certain that a Newman sequence for initial phases will help here. Not had
the time to try it yet.

I uploaded a number of error-correction programs that you may consider.
Let me know if that is of use. See:
(ftp.tapr.org tapr/SIG/hfsig/upload/ecc.zip).

Keep up the good work.

73's

Johan, KC7WW

38. Re: [HFSIG:273] Re: Coding idea for a parallel carrier mode

From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
Date: Wed, 22 Feb 1995 10:41:08 GMT
To: hfsig@tapr.org
Cc: nash@camail.ca.nmp.nokia.com
Message-ID: <199502221042.MAA21873@noknic.nokia.com>
Rendered body type: text/plain

MFSK Synchronisation

The MFSK Mk VI modem (Racal LA1117) uses analogue circuitry to synchronise
to the received signal. The sync circuit consists of two phase locked loops.
The sync signal itself is an idle character consisting of the tones 500Hz and

520Hz. This is the same as phase modulating a 510Hz carrier by +/- 90 degrees since the tones are orthogonal to each other (ie separation = $1/T_s$).

The first PLL has a centre frequency of 510Hz and tracks the phase shift to produce a 10Hz synchronisation signal. This is fed into the second phase locked loop which runs at the symbol rate of 10Hz. When the matched filters detect that 500/520Hz tones are being received, the second PLL is allowed to lock and the sync signal is fed to the quenching circuits of the matched filters (a FET across the feedback capacitor in the matched filter circuit.)

This form of synchronisation is one-shot ie, the idle characters are sent at the start of a message and are not necessarily repeated until the end of the Tx. The stability of the frequency standards in the radios and modem needs to be 0.1ppm for the system to stay locked over 14 hours. In practice, this is actually achieved.

The 33 tone MFSK system uses 10% amplitude modulation applied to the MFSK which is separated by the modem. This is continuous synchronisation but has the disadvantages of increasing the bandwidth, putting special constraints on the AGC of the receiver and being sensitive to amplitude variations (eg fading.)

Modern MFSK modems use Modulation Derived Synchronisation (MDS). This is a DSP technique which enables good synchronisation to be obtained at any point in the Tx without the need for idle characters. This may be appropriate to the OFDM modem. It uses the property that the ratio of the magnitude squared of the strongest peak in an FFT to the sum of the magnitude squared of the other FFT points is a very useful measure of detection confidence and can be used as a soft decision. Typically, it will contain a strong component at the symbol rate. However, the symbol rate information is lost if the same tone is received for two or more symbols. In this case, a digital phase locked loop arrangement is used which is kicked into phase by the confidence estimate signal and used as a flywheel if the signal fades or the same tone is received for consecutive symbols.

For MFSK it works very well with fast locking.

Regards

Adrian G4ZHZ

39. Re: [HFSIG:273] Re: Coding idea for a parallel carrier mode

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 22 Feb 1995 12:24 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <D7B13C9DC0604B44@chopin.ifj.edu.pl>
Rendered body type: text/plain

>Thanks for the explanation - I got it now.
>Another question: could you define "useless" frequency components?

OK. A very first approach was to cut off all frequency bins with power ($\text{power} = I^2 + Q^2$) less than a fixed (user-selectable) threshold. This was cutting well the background noise if the speech signal was significantly stronger than the noise. I noticed you need to move the threshold quite high to completely remove the noise. With moderate thresholds settings you could still hear occasionally very characteristic "bleep, bleep" sounds when single channels were coming above the threshold. I didn't notice much improvement in readability - for weak signals the weaker speech pieces could get cut off completely.

Then I tried to make the cut "smoother": if the bin power was above the threshold I just let it pass. If below I multiplied it by a factor: $\text{bin\_power}/\text{threshold\_power}$. This way nothing is cut off, it's only attenuated proportionally to it's "un-importance".

Then I noticed my radio doesn't pass the audio spectrum in a uniform fashion... I developed a way to estimate the noise level for every frequency bin individually and my threshold was proportional to that noise level. This improved fidelity on AM stations (especially on music). The side effect which I did not expect was that a dead carriers got cut off because they were assumed "noise" - makes some sense by the way :-)

Then I wanted to kill these bleeps... so I tried to correlate the data in time. I required that a given frequency passes the threshold in two (a user-selectable parameter) consecutive FFT samples. I got certain improvement that way, but I am really not sure whether it was worth the effort. Correlating on a longer time scale

could be usefull for CW signals...

I came back to speech/music fidelity again: if a given frequency got triggered I left it open for some time after the trigger so the falling-type sounds (musical instruments) could be passed with least distortions. For even less distortion I was opening the frequency before the trigger (of course you have to use a delay tap for that). A yet another option was that the filter would open the adjacent bins to the triggered ones. All that attributed to fidelity but it let as well more noise to come through, so the final effect was always some sort of trade-off.

The major disadvantage of that automatic background estimation was that the filter needed about 5-10 seconds to adjust itself to new conditions and it was sensitive to fast amplification changes - like when a spike was "pushing up" the AGC. I have some ideas about how to cure these effects but for the moment I got involved in the multi-modem idea :-)

By the way, these games with threshold setting might be usefull for non-coherent type multi-carrier modems...

>I also played around a bit with a simulation program to investigate the >OFDM waveform using raised cosines - the results were not encouraging, >like you indicated. You trade off performance (assuming you have sync) >for the ability to sync/bit-track on-the-fly. I am not sure you want >that when dealing with weak signals. There will be penalties, however, >especially under multipath conditions.

My conclusions are that you have to do longer FFTs: at least 2 times longer and then use every second carrier or better 4 times longer and use every fourth carrier. The unused carriers will be usefull on the receiver side for mis-tune estimation. The simulations I have done tell me that it might be worth to send symbols at a bit lower rate than the maximum one. For example if I use a $\sin(x)/x$ type window I have hardly any frequency crosstalk, and the time crosstalk has a local zero point at about 1.2 the minimum symbol spacing. The other way to do the trade-off is to use say every fifth carrier not every forth, if you do 4 times longer FFTs.

If you stick to the "trivial" FFT aproach with a rectangular window, then a dead carrier somewhere in the receiver passband with affect more or less all the frequency channels (as seen by the receiver). If the carrier is strong enough it would wipe out all the bits - even the strongest ECC code is not going to help you :-)

Paweł

40. [HFSIG] HFSIG@PACCOMM.COM

From: TAPR MAIL MGMT ACCOUNT <taprmail@dingus.n5lyt.datarace.com>
Date: Mon, 20 Feb 1995 19:56:58 -0600 (CST)
To: HFSIG@PACCOMM.COM
Message-ID: <199502210157.TAA13588@dingus.n5lyt.datarace.com>
Rendered body type: text/plain

Dear TAPR Listserv Member:

This is a monthly message to check to see that you are still active on this Mail list and to uncover any bad Internet addresses.

To stay with this mail list, simply DELETE this message. No other action is required. -----

If you want to unsubscribe, send e-mail to listserv@tapr.org with a message of unsubscribe listname, where listname is the name of the list.

Cheers - TAPR

41. Re: [HFSIG:268] Re: Coding idea for a parallel carrier modem

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 22 Feb 1995 17:49:07 -0800
To: JALOCHA@chopin.ifj.edu.pl
Cc: hfsig@tapr.org
Message-ID: <199502230149.RAA20849@servo.qualcomm.com>
Rendered body type: text/plain

>Going to more carriers makes the signaling rate lower (which is good)
>and that simple Hamming code carries less overhead. Just a little table:

Yes, the Hamming code overhead is lower as you increase the blocksize. But since it can correct only one error per block, its effectiveness also decreases with increasing block size.

>I just don't know enough details about RS (or any other) code to write
>the coder/encoder myself :-)

No need. Two years ago, Paul Flaherty, N9FZX, wrote a Reed-Solomon encoder/ decoder routine that he has made freely available. It's available from several archive sites, e.g.,

ftp://labrea.stanford.edu/pub/gnu/ecc-1.2.1.tar.gz
ftp://pdq.coe.montana.edu/pub/mirrors/prep-gnu/ecc-1.2.1.tar.gz

Archie lists several other sites.

Phil

42. Re: [HFSIG:271] Collection of ECC programs

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 22 Feb 1995 18:05:32 -0800
To: FORRERJ@frl.orst.edu
Cc: hfsig@tapr.org
Message-ID: <199502230205.SAA21014@servo.qualcomm.com>
Rendered body type: text/plain

>Phil promised some convolutional code - how about it Phil?

One of these days...I'm thinking of turning it into a Dr. Dobbs' article...

Phil

43. Re: [HFSIG:261] Re: Coding idea for a parallel carrier mode

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 22 Feb 1995 18:08:58 -0800
To: JALOCHA@chopin.ifj.edu.pl
Cc: hfsig@tapr.org
Message-ID: <199502230208.SAA21022@servo.qualcomm.com>
Rendered body type: text/plain

>which window shape shall I use for "sliding window" ?

There are many window shapes. All are compromises. I can cite any number of textbooks that discuss them, do you have good references?

Phil

44. misconfigured mail reflector

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 22 Feb 1995 18:11:56 -0800
To: hfsig@tapr.org
Message-ID: <199502230211.SAA21038@servo.qualcomm.com>
Rendered body type: text/plain

The software that handles the hfsig mailing list is incorrectly configured. It adds a "Reply-To: hfsig@tapr.org" header to the front of every message it relays. This makes it tedious to reply privately to the sender of a message, as I have to edit the destination address manually.

Can whoever maintains the mailing list please look into fixing this? Thanks.

Phil

45. Re: [HFSIG:280] misconfigured mail reflector

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Wed, 22 Feb 95 22:15:03 CST
To: hfsig@tapr.org
Message-ID: <2617@k5yfw.ampr.org>
Rendered body type: text/plain

In message <199502230211.SAA21038@servo.qualcomm.com> you write:
> The software that handles the hfsig mailing list is incorrectly
> configured. It adds a "Reply-To: hfsig@tapr.org" header to the front
> of every message it relays. This makes it tedious to reply privately
> to the sender of a message, as I have to edit the destination address
> manually.

>
> Can whoever maintains the mailing list please look into fixing this? Thanks.
>
> Phil
>
>

Phil,

I believe Lee, N5LYT, maintains it. I will refer this to him. I believe that the actual hosts is dingus.n5lyt.datarace.com and/or dingus.n5lyt.ampr.org (same box).

Walt (k5yfw@sat.n5lyt.ampr.org)

46. RE: [HFSIG:280] misconfigured mail reflector

From: Murphree Michael A <mamurphree@space.honeywell.com>
Date: Thu, 23 Feb 95 08:23:00 PST
To: hfsig <hfsig@tapr.org>
Message-ID: <2F4CB4B7@smtpdos.space.honeywell.com>
Rendered body type: text/plain

>The software that handles the hfsig mailing list is incorrectly
>configured. It adds a "Reply-To: hfsig@tapr.org" header to the front
>of every message it relays. This makes it tedious to reply privately
>to the sender of a message, as I have to edit the destination address
>manually.

It gets worse.... Those of us(?) using Microsoft Mail do not get the headers and can not reply to or even identify the sender unless it is in the From or To fields..

Stuck in the Gates of Hell,
Mike

47. Re: [HFSIG:279] Re: Coding idea for a parallel carrier mode

From: JALOCHA@chopin.ifj.edu.pl
Date: Thu, 23 Feb 1995 12:56 GMT+1
To: hfsig <hfsig@tapr.org>
Message-ID: <A568F1CBC0605D60@chopin.ifj.edu.pl>
Rendered body type: text/plain

>There are many window shapes. All are compromises. I can cite any number of
>textbooks that discuss them, do you have good references?

After re-thinking the problem, my current view is:

On the receiver side the window+FFT works like a FIR filter:
for a given frequency bin we convolute the input signal with
a shape = $\sin(2\pi f t) * \text{window\_shape}(t)$ where f is the frequency
corresponding to this bin: $f = \text{bin\_index} / \text{FFT\_len} * \text{sampling\_rate}$
($\text{bin\_index} = 0 \dots \text{FFT\_len}/2 - 1$).

Now, the problem is "reduced" to finding the FIR shape such that:
- the spectral response is rather flat within the bin's passband
- the spectrum tails going to adjacent bins are minimal
- the crosstalk between successive symbols (on same carrier) is minimal
- the shape is rather short (so we don't have to make long FFTs)
You can't have all at once - certainly compromises are needed.
I believe you can't achieve reasonable performance by doing
the FFT with the "minimal" length - you need to at least double it.

Sounds like an academic (but still non-trivial) problem...
any good student out there ? :-)

Pawel

48. Indian HF modem

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 24 Feb 1995 19:06:10 -0800
To: hfsig@tapr.org
Message-ID: <199502250306.TAA13111@servo.qualcomm.com>
Rendered body type: text/plain

I just got a copy of a very interesting technical report from Bharat Electronics, an Indian company. They did a 2400 bps HF radio modem for the Indian military that uses MIL-STD-188-110 modulation (16 tone

version) plus a (15,9) reed-solomon code over GF(16). On top of that is an interesting hybrid arq/fec scheme. The modem and coder was implemented on a TMS320C30, the arq stuff on a PC.

I've run off 5 copies. I could bring them with me to the TAPR meeting next weekend in St. Louis, for those of you who are coming.

Phil

49. Re: [HFSIG:284] Indian HF modem

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Fri, 24 Feb 1995 23:34:35 PST8PDT
To: hfsig@tapr.org
Message-ID: <2F0B7F64EBA@frl.orst.edu>
Rendered body type: text/plain

Hi Phil,

I'll be interested in seeing the docs. Will appreciate a copy - See you in St. Louis.

73's

Johan KC7WW

50. No Subject

From: BobH@eworld.com
Date: Sat, 25 Feb 95 11:18:46 PST
To: hfsig@tapr.org
Message-ID: <9502251118.tn09091@eworld.com>
Rendered body type: text/plain

Two items of interest:

> To be filed with the growing list of affordable hardware options for DSP-- EZDSP-Lab products from Real Time Signal Processing Inc. (403)250-2407, FAX (403)250-6711.

Stand-alone board with Analog Devices ADSP-2115 and AD1847 Codec for \$99. Price includes PC download utility, Analog Devices' Assembler, Linker and Prom Splitter but not the Simulator. Don't know what debug environment is like. ADSP-2181/AD1847 for \$199.

> Paper I haven't got yet but looks interesting-- "Soft Decision Decoding of Reed-Solomon Codes Using Trellis Methods" IEE PROC-COMM Oct. 94
wm6h

51. Re: [HFSIG:284] Indian HF modem

From: gwyn@paccomm.com (Gwyn Reedy, WIBEL)
Date: Sat, 25 Feb 95 17:22:30 UTC
To: hfsig@tapr.org
Message-ID: <622@paccomm.com>
Rendered body type: text/plain

sounds very similar to the Harris 39 tone modem which used 320c25s I think. Harris moved to 320C50s later on, but had a lot of trouble with early chips.

I'd be very interested in seeing a copy of the report.

C U in St. Louis,
Gwyn

Gwyn Reedy, WIBEL gwyn@paccomm.com WIBEL@amsat.org

+(813) 874-2980 Fax:+(813) 872-8696 BBS: +(813) 874-3078 (V.34)

PacComm Packet Radio Systems, Inc, 4413 N. Hesperides St, Tampa, FL 33614-7618

52. Mail Archives available on ftp.tapr.org

From: Greg Jones <gjones@tenet.edu>
Date: Mon, 27 Feb 1995 00:37:43 -0600 (CST)
To: netsig@tapr.org (NETSIG mailing), bbssig@tapr.org (BBS SIG mailing),
aprssig@tapr.org, hfsig@tapr.org (HF SIG mailing)
Message-ID: <199502270637.AAA15814@Alice-Thurman.tenet.edu>
Rendered body type: text/plain

Mail Archives for the various mail groups are now available on both listserv@tapr.org and via ftp.tapr.org

The mail archives on the listserv is updated daily.

53. Re: [HFSIG:284] Indian HF modem

From: mcdermot@aud.alcatel.com (Tom Mcdermott)
Date: Mon, 27 Feb 1995 07:31:42 -0600
To: hfsig@tapr.org
Message-ID: <95Feb27.073138cst.20494@janus01.aud.alcatel.com>
Rendered body type: text/plain

Phil: I'll also be in St. Louis, and would like a copy if possible.

- Tom, N5EG

54. Re: Indian HF modem

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Mon, 27 Feb 1995 15:38:14 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <199502271438.PAA19390@hades.uet.e-technik.th-darmstad>
Rendered body type: text/plain

Hi!

I also would like to get a copy of that paper. Unfortunately the conference is a bit too far away for me...

I'm watching this group for some time now and I have to thank you all for very interesting information on HF data transmission so far.

I have done a project on spread spectrum data transmission on HF 2 years ago. I had to simulate the modem using the simulation software SPW. For proper testing I implemented the CCIR (Watterson) modell in SPW.

Currently I'm trying to port this channel model to Ptolemy, a CAE tool similar to SPW, but PD software. Ptolemy runs under Linux, so you can use it on a 486.

About the HF simulator: I read trough the mentioned paper on a realtime HF radio channel analyzer (COM-30, pp. 1809-1817) . Why are they using Butterworth filters for the low-pass Gaussian process? Reading through the CCIR report 549 I thought I had to use filters with Gaussian frequency response. Is this just an approximation?

73' Alexander DL8AAU

55. Re: [HFSIG:290] Re: Indian HF modem

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 27 Feb 1995 10:18:48 PST8PDT
To: hfsig@tapr.org
Message-ID: <32B761B3EE3@frl.orst.edu>
Rendered body type: text/plain

Welcome Alexander!

< some lines deleted >

>

> I'm watching this group for some time now and I have to thank you all
> for very interesting information on HF data transmission so far.

> I have done a project on spread spectrum data transmission on HF 2 years
> ago. I had to simulate the modem using the simulation software SPW.
> For proper testing I implemented the CCIR (Watterson) modell in SPW.

I wonder if your simulation also included the noise component, i.e. stochastic + markov burst noise processes in addition to the fading paths? I have a rough idea how that is done, but would like to learn more about it.

> About the HF simulator: I read trough the mentioned paper on a realtime
> HF radio channel analyzer (COM-30, pp. 1809-1817) . Why are they using
> Butterworth filters for the low-pass Gaussian process? Reading through
> the CCIR report 549 I thought I had to use filters with Gaussian frequency
> response. Is this just an approximation?

There is a good reason, just slipped my memory - Hi. I'll get back to you on this one.

73's

Johan, KC7WW

56. Re: [HFSIG:286] No Subject

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 27 Feb 1995 10:19:52 PST8PDT
To: hfsig@tapr.org
Message-ID: <32B7A5348DC@frl.orst.edu>
Rendered body type: text/plain

Bob,

>
> Two items of interest:
>> To be filed with the growing list of affordable hardware options for DSP-
>> EZDSP-Lab products from Real Time Signal Processing Inc. (403)250-2407,
>> FAX (403)250-6711.
>> Stand-alone board with Analog Devices ADSP-2115 and AD1847 Codec for \$99.
>> Price includes PC download utility, Analog Devices' Assembler, Linker and
>> Prom Splitter but not the Simulator. Don't know what debug environment
>> is like. ADSP-2181/AD1847 for \$199.

Thanks for that info Bob - I, for one, are always looking.

> Paper I haven't got yet but looks interesting-- "Soft Decision Decoding of
> Reed-Solomon Codes Using Trellis Methods" IEE PROC-COMM Oct. 94
>

73's

Johan

57. Re: Sound cards

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 28 Feb 1995 08:51:07 PST8PDT
To: hfsig@tapr.org
Message-ID: <3420089300D@frl.orst.edu>
Rendered body type: text/plain

Dear Frode,

I have some rather sad news about the sound cards: called this morning about the whereabouts of the cards and was told that there was no stock left when my order was taken - nobody let me know either. Well, what could I say - dissapointed of course. The alternative now is to get you an Orchid card, such as the one that I got for Adrian for \$99. Would that be acceptable? I still have the cheque, so feel free to change you mind too. Let me know as soon as convenient and I'll do what is necessary.

I apologize for keeping you waiting - wish I could have done more.

73's

Johan, KC7WW

58. HF Channel Simulator

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 28 Feb 1995 09:21:05 PST8PDT
To: hfsig@tapr.org
Message-ID: <34280A72BE9@frl.orst.edu>
Rendered body type: text/plain

Dear Alexander,

<... some lines deleted>

> About the HF simulator: I read trough the mentioned paper on a realtime
> HF radio channal analyzer (COM-30, pp. 1809-1817) . Why are they using
> Butterworth filters for the low-pass Gaussian process? Reading through
> the CCIR report 549 I thought I had to use filters with Gaussian frequency
> response. Is this just an approximation?

The procedure is as follows (this is performed in triplicate):

- 1). Create White Gaussian noise with a given standard deviation as determined by the CCIR conditions that you are simulating.
- 2). Apply a complex low-pass filter to the points (complex) created in (1). A Butterworth response is probably a good choice for having low ripple in the pass band. A two-pole filter is quite adequate.
- 3). Create the Doppler signal (complex) according to CCIR condition.
- 4). Perform the complex mix of (2) + (3).
- 5). Perform the complex mix of (4) and the signal from the tapped delay line.

Note that for step (5) some further processing is required as you have to deal with different sample rates, i.e. the tap gain function at a very low frequency while the delay line is sampled baseband. This tells us that you probably will need additional interpolation filter in the tap-gain path.

At this point, the model has only done multipath fading, the noise component still has to be added. This is a two-component signal, i.e. additive white Gaussian noise in addition there should be a Markov process to simulate burst noise, these however, are now part of a linear model - all the time-variant stuff have been taken care of by then.

Having both the fading path and the noise path in your simulator, one can do S/N performance evaluation in addition to multipath.

Hope this helps a bit,

73's

Johan, KC7WW

March 1995

Original source file: hfsig.9503

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1995-03/TAPR-HF_1995-03.txt

1. Reed-Solomon Package

From: Frode Weierud <frode@dxcern.cern.ch>
Date: Wed, 1 Mar 1995 11:05:29 +0100 (MET)
To: HFSIG TAPR <hfsig@tapr.org>
Message-ID: <Pine.ULT.3.91.950301110059.3464A-100000@dxcern.cern.ch>
Rendered body type: text/plain

Hi all,

I have just uploaded a package of routines for experimenting with Reed-Solomon error correcting codes on ftp.tapr.org, in the tapr/SIG/hfsig/upload directory. The archive is called RSpack.zip.

Here is the content of the RSpack.txt file:

RSpack is a complete package for experimenting with Reed-Solomon error correcting codes. The author of RSpack is:

Jan-Mark Wams.
email: jms@cs.vu.nl
smaile: Jan-Mark Wams
Jephtastraat 67-5
1055 JS Amsterdam
voice: .. (31-20 / 020) 6 81 43 84

Here is a short extract from the readme file:

T H E R S - P A C K A G E

Below is my RS-package. It should run on MS-DOS and UNIX provided a ANSI-C compiler is used (like gcc or turbo C).

It has two main parts:

- i) A generator program.
- ii) A (en/de)-code program.

The first program ('`rsgen.c'`) is a C program that generates a include file for the second program ('`rsbody.c'`). The (en/de)-coding is an example of how the generated file of the generator program could be used. The net effect of a version of the (en/de)-code-program is a filter, that allows one to protect data. For testing purposes, there is a program for altering files, and a program for deleting bytes from a file. These can be used to simulate a bad network.

I have made some small changes to the original Makefile and a few of the source files as describes in the readme.1st file. I have also included a small tutorial on Reed-Solomon codes written by Colin Plumb.

Contributed by Frode Weierud, F/LA2RL
1 March 1995

73 Frode

```
*****
*      Frode Weierud      Phone   :      41 22 7674794      *
*      CERN, SL          Fax     :      41 22 7679185      *
*      CH-1211 Geneva 23  E-mail  :      frode@dxcern.cern.ch *
*      Switzerland                               *
*****
```

2. (no subject)

From: STUART FAWCETT <SJF@rpsl.co.uk>
Date: Wed, 1 Mar 1995 10:18:34 GMT
To: hfsig@tapr.org
Message-ID: <300F4F20E36@mail.rpsl.co.uk>
Rendered body type: text/plain

3. Re: HF Channel Simulator

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
 Date: Wed, 1 Mar 1995 18:58:37 +0100 (MEZ)
 To: hfsig@tapr.org
 Message-ID: <199503011758.SAA17859@hades.uet.e-technik.th-darmstad>
 Rendered body type: text/plain

> The procedure is as follows (this is performed in triplicate):
 >
 > 1). Create White Gaussian noise with a given standard deviation as
 > determined by the CCIR conditions that you are simulating.
 >
 > 2). Apply a complex low-pass filter to the points (complex) created
 > in (1). A Butterworth response is probably a good choice for
 > having low ripple in the pass band. A two-pole filter is quite
 > adequate.
 >
 > ...
 >
 > Johan, KC7WW

The generated Gaussian noise is white in the frequency domain. From the CCIR report and from the Watterson paper it is clear, that the tap-gain function spectrum is the sum of two Gaussian functions of frequency. So I still think you have to use filters with Gaussian frequency response to convert the white noise to the spectrum we need. Furthermore, the CCIR report states that not only the tap-gain function has a Gaussian frequency spectrum with high probability but that a single-pole spectrum is not valid. So a two-pole filter can't be anything more but an approximation.

The filters with Gaussian frequency response can easily be implemented as FIR filters (ok, Butterworth IIR filters would be easier... :-)).

As this is a rather important part of the simulator I think it is still important to talk about this filter...

73' Alexander DL8AAU

```
*-----*
|      Alexander F. Kurpiers      |
|      Technische Hochschule Darmstadt      |
|      Institut fuer Uebertragungstechnik      |
|      Merckstrasse 25      |
|      D-64283 Darmstadt (Germany)      |
|      Voice: +49-6151-162369      |
|      Fax   : +49-6151-165545      |
|      EMail: kurpiers@zeus.uet.e-technik.th-darmstadt.de      |
|      Hamradio: dl8aau@db0kg.#nds.deu.eu      |
*-----*
```

4. Re: [HFSIG:297] Re: HF Channel Simulator

From: mcdermot@aud.alcatel.com (Tom Mcdermott)
 Date: Wed, 1 Mar 1995 14:11:49 -0600
 To: hfsig@tapr.org
 Message-ID: <95Mar1.141122cst.20629@janus01.aud.alcatel.com>
 Rendered body type: text/plain

> > The procedure is as follows (this is performed in triplicate):
 > >
 > > 1). Create White Gaussian noise with a given standard deviation as
 > > determined by the CCIR conditions that you are simulating.
 > >
 > > 2). Apply a complex low-pass filter to the points (complex) created
 > > in (1). A Butterworth response is probably a good choice for
 > > having low ripple in the pass band. A two-pole filter is quite
 > > adequate.
 > >
 > > ...
 > >
 > > Johan, KC7WW
 >
 > The generated Gaussian noise is white in the frequency domain. From the CCIR
 > report and from the Watterson paper it is clear, that the tap-gain function
 > spectrum is the sum of two Gaussian functions of frequency. So I still
 > think you have to use filters with Gaussian frequency response to convert
 > the white noise to the spectrum we need. Furthermore, the CCIR report

> states that not only the tap-gain function has a Gaussian frequency
> spectrum with high probability but that a single-pole spectrum is not
> valid. So a two-pole filter can't be anything more but an approximation.
>
> The filters with Gaussian frequency response can easily be implemented as
> FIR filters (ok, Butterworth IIR filters would be easier... :-)).
>
> As this is a rather important part of the simulator I think it is still
> important to talk about this filter...
>
> 73' Alexander DL8AAU
>

Alexander: I believe what the CCIR report requires is that the doppler frequency have a gaussian-distributed probability density function (PDF). This is not the same as the amplitude response of the filter. The gaussian PDF is generated by a gaussian random number generator. Since gaussian noise has a flat frequency response, a low pass filter obviously distorts the PDF. But a gaussian-amplitude LPF does not mean the the PDF of the output of the LPF is gaussian, because the LPF would have to be flat (ie no filter) to have gaussian PDF. Remember, gaussian PDF -> white noise (spectally flat, not shaped), and not-flat spectrum -> non-gaussian.

So, the filter function does not alter the statistics of the distribution in a readily recognizable way. As the filter bandwidth changes, the standard-deviation of the distribution changes, since we are changing how the random numbers are being averaged -- this is equivalent to saying that the energy in the signal is changed, since RMS = Std. Dev.

The math behind linear amplitude transformation of PDF's, and the resulting PDF is not readily available in the EE literature (well, I haven't seen it anyway). However, the math behind the summation of gaussian-distributed random numbers is well documented. In amplitude, the sum of multiple gaussian numbers with non-zero mean leads to a Rayleigh distribution (PDF) of the resultant amplitude. So this corresponds to the requirement that the path component have Rayleigh fading.

How often the fade occurs is related to the doppler frequency. So a wider low pass filter yields faster fading, equivalent to a larger std deviation of the doppler frequency. The sum of gaussian-distributed complex random numbers generates a random phase with a uniform distribution, and this matches the requirement that the phase of the path component be uniformly distributed.

So, even without filtering, the doppler is gaussian-distributed. The purpose of the filter then is just to set the doppler rate, so we can control it for simulating different band conditions.

- Tom, N5EG

ps: I have no training in the area of stochastic processes, someone who has been there before can give us a discourse on what spectral filtering does to PDF's! It is a difficult job to mentally keep separate amplitude spectrum and PDF.

5. RE: [HFSIG:298] Re: HF Channel Simulator

From: "Richard W. D. Booth" <rick@itron-ca.com>
Date: Wed, 1 Mar 95 14:53:22 PST
To: hfsig@tapr.org
Message-ID: <Chameleon.4.01.950301145609.rick@rick.itron-ca.com>
Rendered body type: text/plain

Tom,
The output of a linear filter driven by a gaussian random process is still a gaussian random process...the spectrum (or autocorrelation function, if you will) is modified by the properties of the filter so the correlation between samples is modified.
Rick
w6nzk
>
>> > The procedure is as follows (this is performed in triplicate):
>> >
>> > 1). Create White Gaussian noise with a given standard deviation as
>> > determined by the CCIR conditions that you are simulating.
>> >
>> > 2). Apply a complex low-pass filter to the points (complex) created
>> > in (1). A Butterworth response is probably a good choice for
>> > having low ripple in the pass band. A two-pole filter is quite

```

>> > adequate.
>> >
>> > ...
>> >
>> > Johan, KC7WW
>>
>> The generated Gaussian noise is white in the frequency domain. From the CCIR
>> report and from the Watterson paper it is clear, that the tap-gain function
>> spectrum is the sum of two Gaussian functions of frequency. So I still
>> think you have to use filters with Gaussian frequency response to convert
>> the white noise to the spectrum we need. Furthermore, the CCIR report
>> states that not only the tap-gain function has a Gaussian frequency
>> spectrum with high probability but that a single-pole spectrum is not
>> valid. So a two-pole filter can't be anything more but an approximation.
>>
>> The filters with Gaussian frequency response can easily be implemented as
>> FIR filters (ok, Butterworth IIR filters would be easier... :-) ).
>>
>> As this is a rather important part of the simulator I think it is still
>> important to talk about this filter...
>>
>> 73' Alexander DL8AAU
>>
>
>
> Alexander: I believe what the CCIR report requires is that the doppler
> frequency have a gaussian-distributed probability density function (PDF).
> This is not the same as the amplitude response of the filter. The gaussian
> PDF is generated by a gaussian random number generator. Since gaussian
> noise has a flat frequency response, a low pass filter obviously distorts
> the PDF. But a gaussian-amplitude LPF does not mean the the PDF of the
> output of the LPF is gaussian, because the LPF would have to be flat
> (ie no filter) to have gaussian PDF. Remember, gaussian PDF -> white noise
> (spectrally flat, not shaped), and not-flat spectrum -> non-gaussian.
>
> So, the filter function does not alter the statistics of the distribution in
> a readily recognizable way. As the filter bandwidth changes, the
> standard-deviation of the distribution changes, since we are changing how
> the random numbers are being averaged -- this is equivalent to saying that the
> energy in the signal is changed, since RMS = Std. Dev.
>
> The math behind linear amplitude transformation of PDF's, and the resulting
> PDF is not readily available in the EE literature (well, I haven't
> seen it anyway). However, the math behind the summation of
> gaussian-distributed random numbers is well documented. In amplitude,
> the sum of multiple gaussian numbers with non-zero mean leads to a Rayleigh
> distribution (PDF) of the resultant amplitude. So this corresponds to the
> requirement that the path component have Rayleigh fading.
>
> How often the fade occurs is related to the doppler frequency. So a
> wider low pass filter yields faster fading, equivalent to a larger
> std deviation of the doppler frequency. The sum of gaussian-distributed
> complex random numbers generates a random phase with a uniform distribution,
> and this matches the requirement that the phase of the path component be
> uniformly distributed.
>
> So, even without filtering, the doppler is gaussian-distributed. The purpose
> of the filter then is just to set the doppler rate, so we can control it
> for simulating different band conditions.
>
>
> - Tom, N5EG
>
>
> ps: I have no training in the area of stochastic processes, someone who
> has been there before can give us a discourse on what spectral filtering
> does to PDF's! It is a difficult job to mentally keep separate amplitude
> spectrum and PDF.
>
>

```

6. Re: HF Channel Simulator

```

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Thu, 2 Mar 1995 09:23:34 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <199503020823.JAA09206@hades.uet.e-technik.th-darmstad>
Rendered body type: text/plain

>
> Tom,

```

> The output of a linear filter driven by a gaussian random process is still a gaussian
> random process...the spectrum (or autocorrelation function, if you will) is modified by the
> properties of the filter so the correlation between samples is modified.
> Rick
> w6nzk
> >

Thank you, that's what I wanted to reply when your mail arrived here...

> >
> > Alexander: I believe what the CCIR report requires is that the doppler
> > frequency have a gaussian-distributed probability density function (PDF).
> > This is not the same as the amplitude response of the filter. The gaussian
> > PDF is generated by a gaussian random number generator. Since gaussian
> > noise has a flat frequency response, a low pass filter obviously distorts
> > the PDF. But a gaussian-amplitude LPF does not mean the the PDF of the
> > output of the LPF is gaussian, because the LPF would have to be flat
> > (ie no filter) to have gaussian PDF. Remember, gaussian PDF -> white noise
> > (spectrally flat, not shaped), and not-flat spectrum -> non-gaussian.

I know that a Gaussian distributed random variable (so the PDF is gaussian-distributed) has nothing to do with the power spectral density. In our case it seems to be clear:
(Watterson, Experimental Confirmation of an HF Channel Model):

Statistical specifications for the tap-gain function involved three hypotheses: 1.) that each tap-gain function is a complex-Gaussian process that produces Rayleigh fading
2.) that the tap-gain functions are independent
3.) that each tap-gain function has a spectrum that in general is a sum of two Gaussian functions of FREQUENCY, one for each magnetoionic component.

1. is related to the pdf and without 2. we would have trouble implementing the model. But 3. says clearly, what the psd of the tap-gain functions have to look like.

> > So, the filter function does not alter the statistics of the distribution in
> > a readily recognizable way. As the filter bandwidth changes, the
> > standard-deviation of the distribution changes, since we are changing how
> > the random numbers are being averaged -- this is equivalent to saying that the
> > energy in the signal is changed, since RMS = Std. Dev.

I was trying to calculate the RMS and found an equation by numeric integration. You will have to calculate it, otherwise you have to estimate the mean signal power at the point, where Gaussian noise is added to the fading signals (at the output of the HF simulator).

> > The math behind linear amplitude transformation of PDF's, and the resulting
> > PDF is not readily available in the EE literature (well, I haven't
> > seen it anyway). However, the math behind the summation of
> > gaussian-distributed random numbers is well documented. In amplitude,
> > the sum of multiple gaussian numbers with non-zero mean leads to a Rayleigh
> > distribution (PDF) of the resultant amplitude. So this corresponds to the
> > requirement that the path component have Rayleigh fading.

As stated above, linear filtered of Gaussian distributed random variables will still be Gaussian distributed. You can think of the filtering process as a convolution with the impulse response of the filter. If you do this convolution with samples of the random variable (discrete time filtering), it's nothing more than multiplying the samples with constants and summing the result. As you said, the sum of any number of Gaussian distributed numbers is Gaussian ... :-)
If you do the filtering continuously, the summing will be an integration.

About literature, try Papoulis, "Probability, Random Variables and Stochastic Processes". The book I'm using here is in German, so it is probably of no use for you.

> > How often the fade occurs is related to the doppler frequency. So a
> > wider low pass filter yields faster fading, equivalent to a larger
> > std deviation of the doppler frequency. The sum of gaussian-distributed
> > complex random numbers generates a random phase with a uniform distribution,
> > and this matches the requirement that the phase of the path component be
> > uniformly distributed.

Not only the sum of Gaussian-distributed complex numbers has a random phase...
The phase of any Gaussian-distributed complex number with zero mean Gaussian distributed real and imaginary parts is uniform distributed and

the amplitude is Rayleigh distributed.

```
> >
> > So, even without filtering, the doppler is gaussian-distributed. The purpose
> > of the filter then is just to set the doppler rate, so we can control it
> > for simulating different band conditions.
> >
> >                                     - Tom, N5EG
> >
> >
> > ps: I have no training in the area of stochastic processes, someone who
> > has been there before can give us a discourse on what spectral filtering
> > does to PDF's! It is a difficult job to mentally keep separate amplitude
> > spectrum and PDF.
> >
```

73' Alexander DL8AAU

7. Re: [HFSIG:299] Re: HF Channel Simulator

From: mcdermot@aud.alcatel.com (Tom Mcdermott)
Date: Thu, 2 Mar 1995 07:47:58 -0600
To: hfsig@tapr.org
Message-ID: <95Mar2.074733cst.20648@janus01.aud.alcatel.com>
Rendered body type: text/plain

```
> Tom,
> The output of a linear filter driven by a gaussian random process is still a gaussian
> random process...the spectrum (or autocorrelation function, if you will) is modified by the
> properties of the filter so the correlation between samples is modified.
> Rick
> w6nzk
```

Rick: I agree. The power spectral density is, I think, the Fourier transform of the autocorrelation function, which for zero-mean gaussian process results in gaussian PSD. Since the gaussian PSD rolls-off pretty fast in amplitude vs. frequency already, the filter shape doesn't mess up the PSD too much, as long as it is reasonably square-ish (ie several poles). Sure, the shape is changed, but the error is fairly small, and is ignored for a multi-pole filter.

So, filtering the white-noise signal results in another (almost) white noise signal of lower bandwidth, but is still has pretty much the same PSD shape except with a lower std. deviation. (PSD is equivalent to a PDF, I believe).

- Tom, N5EG

8. Re: HF Channel Simulator

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Thu, 2 Mar 1995 15:56:44 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <199503021456.PAA20626@hades.uet.e-technik.th-darmstad>
Rendered body type: text/plain

```
>
> Rick: I agree. The power spectral density is, I think, the Fourier
> transform of the autocorrelation function, which for zero-mean gaussian
> process results in gaussian PSD. Since the gaussian PSD rolls-off pretty
> fast in amplitude vs. frequency already, the filter shape doesn't mess up
> the PSD too much, as long as it is reasonably square-ish (ie several poles).
> Sure, the shape is changed, but the error is fairly small, and is ignored
> for a multi-pole filter.
```

You can't tell the psd (power spectral density) from the pdf (probability density function), which is a measure for the amplitude distribution of a random variable. The psd is related to the autocorrelation function, it is indeed the Fourier transform of it.

I.e. if the Gaussian distributed random variable is uncorrelated, the psd is a constant and thus the noise is called white. Filtering this white noise results in a new more or less correlated random variable whose pdf is still Gaussian distributed.

```
>
> So, filtering the white-noise signal results in another (almost) white
> noise signal of lower bandwidth, but is still has pretty much the same PSD
> shape except with a lower std. deviation. (PSD is equivalent to a PDF,
> I believe).
>
```

>
> - Tom, N5EG

73' Alexander DL8AAU

9. Re: HF Channel Simulator

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Wed, 8 Mar 1995 09:10:30 -0600
To: hfsig@tapr.org
Message-ID: <95Mar8.091039cst.20587@janus01.aud.alcatel.com>
Rendered body type: text/plain

I have performed some analysis on the question of the shape of the tap-gain functions. I now understand Alexander's original question. My previous statements about the filter shape are not correct. Here's what I did, and the results.

1. Generated a real-valued random number stream using gaussian-distributed random number, and calculated power spectral density (PSD) using two methods, Fourier transform of autocorrelation integral, and using Welch's periodogram. Both give white PSD (ie: flat PSD vs. frequency).
2. Generated a complex-valued random number stream, with I- and Q- components generated by two independent gaussian-distributed random number sequences. Using both methods to compute PSD, as above, the PSD was white. The amplitude PDF is Rayleigh, the phase PDF is uniform.
3. Designed an IIR LPF filter, 2nd order Butterworth, with corner frequencies of 0.1, and 0.25. Then filtered the random sequence generated in step 2 with this filter.
4. Calculated the PSD of the filtered signal from step 3 with both methods. Each gives a lowpass PSD resembling the Butterworth filter response curve. This corresponds to the tap-gain spectrum, if one looks at the double-sided spectrum.

Notes from observation of the filter shape of the Butterworth responses:

1st order filter has poor roll-off, the shape does not look like a gaussian roll-off. Lot of higher frequency energy. This results in tap-gain spectrum that does not look gaussian, ie: too much energy too far away from the center frequency -> pessimistic results.

2nd order Butterworth LPF gives very nice roll-off characteristic, closely matching gaussian response. It's slightly distorted at low frequency, good match in transition region and cut-off region. If one were to generate the double-sided spectrum, this would give a result matching the gaussian spectrum roll-off, except that the part in the very center would be squashed down a little (flat across the top, instead of bullet-nosed).

3rd order Butterworth LPF gives a spectrum with a too-sharp roll-off. This would be too rectangular- as compared to the gaussian curve.

So, the conclusion I would reach is that the 2nd order Butterworth is a pretty good approximation to the gaussian tap-gain spectrum shape.

- Tom, N5EG

PS: This discussion over the last week on hfsig has been extremely educational. Thanks!

10. Annual meeting update

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 9 Mar 1995 14:03:44 PST8PDT
To: HFSIG@tapr.org
Message-ID: <41F3CFA3CF9@frl.orst.edu>
Rendered body type: text/plain

Hi All,

Thought that I would welcome all those that joined us after the TAPR annual meeting.

For those that did not have the opportunity to attend - it was quite interesting. Was my first time. I was taken with the richness in diversity of fields of interests and capabilities of participants.

There was a lot of emphasis on DSP. Bob Stricklin made a valiant effort to stimulate the discussion in the DSP developer's symposium, however, it quickly evolved into a "wishlist for digital radio features". In my humble opinion, the opportunity was lost to bring together those that are actually developing DSP applications. It was a large audience, evidently with lots of different expectations, just my opinion that there were too many with other interests that overwhelmed the purpose of the meeting.

Tom McDermitt, N5EG, made an outstanding presentation on channel characterization - I certainly enjoyed his approach and learned a few things. I am looking forward to Tom's book.

Saturday afternoon, we had an HFSIG meeting that was attended by 15 - 20 folks. This meeting paralleled with another on TCP/IP, so guess we had the *REAL* HF digital enthusiasts together. I presented a summary of recent activities that included: the work on the HF channel simulator and alternate modulation schemes. I indicated that our efforts are focussed on a) methods for high speed communications suitable for networking applications, and b) low speed, very robust communications to operate in adverse conditions. There is a distinction between these. I also discussed some of the results on improving crest factor. It was evident that the contributions of this SIG was followed by many, and with interest. Thanks to all of you for making this possible.

The dinner presentation was an energetic talk by Paul Schuch on SETI.

Sunday morning was time for Phil Karn's, KA9Q, exceptionally good presentation on error-control coding. It was an enjoyable experience, non-mathematical, however, with enough depth to get the point across. The message was that there is no free lunch - everything is a compromise, the main tradeoffs being bandwidth and power. Block codes work better when block size is larger. Convolutional codes are more computationally expensive, however, yields somewhat better coding gain. Softening up the decisions at the discriminator helps a lot. At its simplest, erasure coding is a start. However, multiple levels of decision codes does wonders. The combination of inner/outer codes was shown to be the ultimate, however, probably also the most demanding as far as computations are concerned. Phil's talk contained numerous interesting asides of how professionals do it on the deep space projects, i.e., how to get the message across when the spacecraft's big antenna is not working as it should.

Sunday afternoon, Bob Stricklin, and Tom McDermitt presented a wealth of useful information for aspiring DSP-93 programmers. It was a lot of fun listening to questions and answers - everyone got their money's worth.

Personally, I thought that it was a worthwhile experience. The board, especially Greg Jones, deserves credit for a job well done. The organizers also did an outstanding job to make us all feel welcome. It was a wonderful auditorium and everything ran like clockwork.

Thanks all,

73's

Johan, KC7WW

11. info

From: techman@ace.com (Techman)
Date: Fri, 10 Mar 95 02:33:55 EST
To: hfsig@tapr.org
Message-ID: <9503100233.0.UUL1.3#25274@ace.com>
Rendered body type: text/plain
subscribe

12. PacTOR II?

From: faunt@netcom.com (Doug Faunt N6TQS 510-655-8604)
Date: Fri, 10 Mar 1995 09:15:42 -0800
To: hfsig@tapr.org
Message-ID: <199503101715.JAA20975@netcom2.netcom.com>
Rendered body type: text/plain

Are there any references available on PacTOR II?
How does it differ from PacTOR?
73, doug

13. Re: [HFSIG:306] PacTOR II?

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Fri, 10 Mar 1995 15:22:50 -0600
To: hfsig@tapr.org
Message-ID: <95Mar10.152307cst.20555@janus01.aud.alcatel.com>
Rendered body type: text/plain

> Are there any references available on PacTOR II?
> How does it differ from PacTOR?
> 73, doug
>

The ADRS Journal for Jan/Feb/Mar has information on PacTOR II.
I have Jan and Feb articles - Jan talks about two-tone DPSK and convolutional coding. Feb issue says little useful. Mar issue is supposed to say more, but am waiting for it.

So far, it's pretty vague about the exact format. Requires unit with a 56000 series DSP and a 68000 series general microprocessor.

I'm guessing it will say that it is two-tone DBPSK 200-baud per tone, with rate=1/2 k=9 convolutional coding, and raised-cosine baseband filtering. But we'll have to wait and see.

- Tom, N5EG

14. Thoughts on HF channel simulation

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 10 Mar 1995 13:40:10 -0800
To: hfsig@tapr.org
Message-ID: <199503102140.NAA00427@servo.qualcomm.com>
Rendered body type: text/plain

Everything I read and hear says that HF channels are really hard to model and simulate realistically. But a repeatable simulation is really important to have if you want to do controlled tests and comparisons.

Would it be possible to probe a real HF channel, record its response digitally, and then use that information in offline simulations that could be replayed as often as desired?

Broadband impulses would make good channel probing signals. Where's the woodpecker now that we need it? :-)

Phil

15. RE: [HFSIG:308] Thoughts on HF channel simulation

From: "Richard W. D. Booth" <rick@itron-ca.com>
Date: Fri, 10 Mar 95 19:23:32 PST
To: hfsig@tapr.org
Message-ID: <Chameleon.4.01.950310193656.rick@rick.itron-ca.com>
Rendered body type: text/plain

>Everything I read and hear says that HF channels are really hard to
>model and simulate realistically. But a repeatable simulation is
>really important to have if you want to do controlled tests and
>comparisons.

>
>Would it be possible to probe a real HF channel, record its response
>digitally, and then use that information in offline simulations that
>could be replayed as often as desired?

>
>Broadband impulses would make good channel probing signals. Where's
>the woodpecker now that we need it? :-)

>
>Phil
>

Good question but may have some difficulties:

1. Broadband impulses are a good way to make the measurements as long as the receiving "bandwidth" is large enough so that it does not distort the measurements. This includes the antenna and front end filters. Not to mention the legal issues...hmmm
2. Narrow band or CW probing has the following problems.
Presumably the amplitude and phase of the incoming signal could be tracked if the receiver LOs were stable enough. Measurement of doppler shift would require knowing the stability at both the transmit and receive ends...
3. Given the above CW measurement could be made then the results must be extended over the signal bandwidth of interest. Assume a flat fading model where the

measurement is assumed to apply to all frequencies in the band of interest?
Multitone probes could be used and the different phase and amplitude characteristics
used to synthesize a time varying filter. Sounds tough.

No doubt a repeatable model is necessary. This must be available to anyone interested
in developing both hardware and software for the solution of the problem. Fixing the
"standard" channel model is likely the only way non subjective evaluations will be
achievable...somehow the BS has got to be separable from the facts!!!

Rick Booth
w6nzk

16. RE: Thoughts on HF channel simulations

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Sat, 11 Mar 1995 12:33:04 +0100 (MEZ)
To: hfsig@tapr.org
Message-ID: <199503111133.MAA16623@hades.uet.e-technik.th-darmstad>
Rendered body type: text/plain

Hi!

This kind of research was done in the early 70's by Watterson et. al. and
lead to a widely accepted model which is described in the CCIR report
549-3 (1990). The original paper is:

Watterson, C.C., Juroshek, J.R. and Bensema, W.D., "Experimental
confirmation of an HF channel model", IEEE Trans. Comm. Tech., COM-18,
1970, pp. 792-803.

This model covers ionospheric transmissions with discrete Rayleigh fading
paths and is found to be very realistic.

73' Alexander

17. Digital Test Files

From: bobs@tcet.unt.edu (Bob Stricklin)
Date: Sat, 11 Mar 95 11:25:14 CST
To: hfsig@tapr.org
Message-ID: <9503111725.AA148471@tcet.unt.edu>
Rendered body type: text/plain

It should be possible to code the Waterson model in such a way that a
DSP unit would produce a standard test pattern. This test pattern could
then be transmitted and recorded by a second HF station equipped with
DSP so the pattern can be digitally recorded and stored. Then the file
could be placed on an FTP site as a test data block. I think this is
what Phil wanted. I have been thinking about the same thing. Actually I
would like to have a collection of digitally recorded files for all the
typical modulation or encoding methods used today. Then you could do
modeling and testing of filter code in EXCEL.

The TAPR DSP-93 should work well in this application.

Bob Stricklin, N5BRG

18. unsubscribe

From: fmitche@rdcclink.rd.qms.com
Date: Sun, 12 Mar 95 16:29:59 CST
To: netsig@tapr.org, bbssig@tapr.org, hfsig@tapr.org
Message-ID: <9502127950.AA795054599@rdcclink.rd.qms.com>
Rendered body type: text/plain

unsubscribe fmitche@rd.qms.com
subscribe fmitche@maf.mobile.al.us

19. HF Simulations

From: wtb@partyline.org
Date: Sun, 12 Mar 95 19:07:23 -0600
To: hfsig@tapr.org
Message-ID: <9503121907.0QV4E00@partyline.org>
Rendered body type: text/plain

QEX, Dec '94, page 9, KE4BAD discusses two HF simulations you might
want to consider. Bill NOLIK 6219146
reply wtb@partyline.org

20. HFSIG Quaterly report

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 13 Mar 1995 08:27:59 PST8PDT
To: HFSIG@tapr.org
Message-ID: <479A6CA084F@frl.orst.edu>
Rendered body type: text/plain

Hi All,

Here is a copy of HFSIG quaterly report. Please let me know of any typo's, misspellings or items of interest that I missed else that is how it goes into PSR.

73's

Johan

++++
HFSIG QUARTERLY SUMMARY: MARCH 1995

Judging from comments expressed at the recent TAPR annual meeting, the activities of HFSIG are followed by many and found to contain interesting and useful material. HF digital is one instance where technical advances and its adoption by the amateur community, can be made with relative ease. Recent examples of this are CLOVER II, G-TOR, and PACTOR II. Many fields in amateur radio do not allow this degree of flexibility in adopting new ideas and standards. The challenge of the HF channel offers a unique opportunity to explore and bring together a mix of theory, hardware, and software. Thanks to all your contributions that makes this possible.

Activities on HFSIG included several topics of interest:

- HF Channel Simulator,
- HF Modem Technology,
- Error-Control Coding.

HF Channel Simulator

An HF channel simulator is an essential tool for HF modem evaluation. For example, comparing TNC's or testing and tuning algorithms. HFSIG's simulator effort has been based on a model of HF propagation developed by Watterson et. al. (1). Simulating the HF channel is a very complex subject - computer models such as IONCAP simulates the nature of the ionosphere from input parameters such as sunspot data, day and time for the year, frequency of operation, also magnetic and geographical location. For our purpose, however, a much simpler approach is followed. A bandwidth-limited, stationary model with operational parameters defined by CCIR recommendations (2) is used. This model is based on work by the Watterson group using actual on-the-air broadband transmissions and found statistically accurate. This is a well-known method used for evaluation and testing purposes. For a description of an actual implementation of such a model, please see reference (3).

Recently, the interpretation of how to generate appropriate fading functions was discussed by Tom McDermott and Alexander Kurpiers. These functions have to represent two independent complex bivariate Gaussian ergodic random processes, each with zero mean and independent real and imaginary components with equal R.M.S. values that produce Rayleigh fading.

Contributors for this discussion:

Rick Whiting, W0TN,
Barry Buelow, W0RJT,
Eric Silbaugh, N2NNP,
Glen Worstell, KG0T,
Rick Booth, W6NZK,
Jon Bloom, KE3Z,
Hugh Shane, N7UAX,
Tom McDermott, N5EG,
Alexander Kurpiers, DL8AAU.

HF Modem Technology

Two objectives are evident: A) High speed modems for eventual application over HF-based networks. B) Robust modems for use under adverse conditions. These two objectives possibly require different approaches and tradeoffs have to be made.

Several promising approaches have been proposed. Orthogonal signaling schemes were suggested by Alan Bloom and Phil Karn. These schemes define multiple signaling channels spaced apart in frequency such that these are essentially independent. Occupied bandwidth varies depending on the number of channels and the symbol rate.

The family of MIL-STD-188, 16 and 39 parallel-tone modems, are good examples of multi-tone modems that are already available as commercial products, while ITA21, 6-tone, and ITA52, 12-tone Piccolo modems are examples of implementation of the MFSK approach. Ongoing work by Paul Russell and Pawel Jalocho continues the parallel approach, whereas Adrian Nash is pursuing the MFSK approach. Extensive literature is available on MFSK - reference (4) is a good place to start.

Waveform synthesis for multi-tone signals was reviewed to reduce extreme amplitude situations, i.e., noise-like spikes. Eric Silbaugh has been looking into these using computer simulation. Barry McLarnon pointed us to appropriate literature on the subject. Please refer to reference (5) for further details.

Contributors in this discussion included:

Rick Whiting, W0TN,
Walt DuBose, K5YFW,
Phil Karn, KA9Q,
Hugh Shane, N7UAX,
Barry McLarnon, VE3JR,
Alan Bloom, N1AL,
Adrian Nash, G4ZHZ,
Charles Brain, G4GU0,
Kok Chen, AA6TY,
Rolf Sommerhalder, HB9CWP,
Frode Weierud, F/LA2RL,
Pawel Jalocho, (S53??),
Paul Russell (non-ham).

Error-Control Coding

The purposes of error-control coding for future HF modems are evident - dealing with fading, burst errors, interference and power considerations. Phil Karn has been a regular contributor and explained how best to employ block, convolutional, and combined clock/convolutional methods. Pawel Jalocho suggested a novel Hamming code for use with his experimental multi-tone modem. Phil has promised to contribute his efforts of an advanced convolutional coder that would ideally be suited for our future HF digital work. We are looking forward to this.

Contributors in this discussion:

Phil Karn, KA9Q,
Pawel Jalocho.

General

A growing library of code examples is available in the HFSIG upload area:

<ftp://tapr.org/tapr/SIG/hfsig/upload>

Example code for the HF simulator, experimental parallel modem, and error-control codes are available. In addition, DSP code for AMTOR/PACTOR using a DSP sound card, also W9GR ported denoising code is available for experimental purposes.

Important notice:

As scribe/HFSIG chair person, this probably will be my last contribution for a while. Life's realities, in my case, my Ph.D work, has required that most everything else must be put on hold. HFSIG thus require assistance in this regard. Please contact Greg

if you like to volunteer to help out. This is a very capable and nice group of people to work with.

Thanks to each and all for participating, good luck, and keep up the good work.

Johan Forrer
(KC7WW)

References:

- 1)
Watterson, C. C., J.R. Jurosheck, and W.D Bensema. "Experimental Confirmation of an HF Channel Model". IEEE Trans. on Comm. Tech. Vol COM-18(3). 1970.
- 2)
CCIR Recommendation 520-1. "Use of High Frequency Ionospheric Simulators".
- 3)
Ehrman, L., L.B. Bates, J.F. Eschle, and J.M. Kates. "Real-Time Software Simulation of the HF Radio Channel". IEEE Trans. on Comm. Tech. Vol COM-30(8). 1982.
- 4)
Ralphs, J.D. "Principles and Practice of Multi-Frequency Telegraphy". Peter Peregrinus Ltd.
- 5)
Boyd, S. "Multitone Signals with Low Crest Factor". IEEE Trans. on Circuits and Sys. Vol CAS-33 (10). 1986.

21. HF Channel simulator

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 13 Mar 1995 08:43:26 PST8PDT
To: HFSIG@tapr.org
Message-ID: <479E8F363CF@frl.orst.edu>
Rendered body type: text/plain

Greetings,

I noticed some recent activities on the HF Simulator topic - good to hear the new ideas.

My \$0.02: I think I hear three different things.

1) Bob needs a standard "test recording" of different signals for purposes of tuning and developing filters.

2) Phil (I think) is interested in knowing the channel's actual transfer function. Knowing this, one may use adaptive techniques. This is not easy as the channel is very dynamic, but I can see that if a form of training sequence is sent ahead of a data transmission by the ISS, and the IRS does the proper analysis, it may signal this back to the ISS for taking appropriate action.

3) Modem evaluation: This is where we want to actually take a baseband signal from a modem under test, play all the tricks that the channel would do to the signal (multipath distortion, flat/selective fading, etc) and pass this modified signal to another modem to decode. I think it is understood that this cannot be done with a pre-recorded data signal. You would want to have the ability to signal message content at will such to obtain a good statistical representative sample of how well the modem performs.

The Watterson model that we are working on will perform (3) above. I do, however, see that (1) and (2) also have merit.

If someone want to take this on, how about start making recordings in ".wav" format and upload it. This way we can can play with it a little.

Hope this helps.

73's

Johan, KC7WW

22. Re: Digital Test Files

From: "Andries Herholdt" <aherhld@iaccess.za>
Date: Mon, 13 Mar 1995 21:47:16 +0000
To: hfsig@tapr.org
Message-ID: <m0roGGg-0007LdC@minnie.iaccess.za>
Rendered body type: text/plain

On Sat, 11 Mar 1995 , Bob Stricklin wrote :

> what Phil wanted. I have been thinking about the same thing. Actually I
> would like to have a collection of digitally recorded files for all the
> typical modulation or encoding methods used today.

Perhaps you are familiar with Klingenfuss Publication's CD's ? They are CD recordings of many typical HF modulation types including many digital ones. Perhaps one could convert them to data format somehow. The most elegant method would be to write a program (Visual Basic?) to read the data straight off the disk using a CDROM. A less elegant way might be to record it straight off play it back through a sound card and get hold of the data there. Of course, Klingenfuss would have to give their blessing to such an endeavour !

One of my clients (I am a freelance engineer) have done what you suggest; they have a library on 1Gbyte laser disks of many signals recorded down here (za = South Africa), which they use for comms research. They might even be persuaded to part with it, but probably not for free :((nothing in it for me BTW).

Perhaps I will be forgiven for slipping in a question at this point that is slightly off the topic ? At the moment I am working on some data decoding software and one of the topics that came up was a 32-tone version of MFSK that Klingenfuss calls CIS Piccolo and apparently also is known as CROWD36. I can manage to extract the raw symbols, but of course the alphabet is the main problem. I was wondering : does anyone have any info on this, or could you point me in the direction of an information source ?

Thanks in advance,
Andries.

Andries Herholdt
aherhld@iaccess.za

23. access

From: wtb@partyline.org
Date: Mon, 13 Mar 95 19:19:37 -0600
To: hfsig@tapr.org
Message-ID: <9503131919.0R5FQ00@partyline.org>
Rendered body type: text/plain

help.
wtb@partyline.org

24. HF simulation

From: wtb@partyline.org
Date: Mon, 13 Mar 95 19:19:36 -0600
To: hfsig@tapr.org
Message-ID: <9503131919.0R5F400@partyline.org>
Rendered body type: text/plain

Can someone tell me were to find out what a .wav file is? How
is it put together and how is it played from the computer?:
Bill N0lik reply wtb@partyline.org

25. Re: [HFSIG:316] Re: Digital Test Files

From: Bob Stricklin <bstrick@tenet.edu>
Date: Mon, 13 Mar 1995 23:05:42 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199503140505.XAA16718@Leslie-Francis.tenet.edu>
Rendered body type: text/plain

According to Andries Herholdt:
>
>

Thanks for your comments. I think it will be easier to make my own files. Sounds like the others may be a little hard to find. Also I have a better idea what is required to generate the files from a radio with the DSP-93.

> Perhaps I will be forgiven for slipping in a question at this point
> that is slightly off the topic ? At the moment I am working on some
> data decoding software and one of the topics that came up was a 32
> -tone version of MFSK that Klingenfuss calls CIS Piccolo and
> apparently also is known as CROWD36. I can manage to extract the raw
> symbols, but of course the alphabet is the main problem. I was
> wondering : does anyone have any info on this, or could you point me
> in the direction of an information source ?

Sorry, I can not help you on this.

Bob Stricklin, NSBRG

26. Re: [HFSIG:316] Re: Digital Test Files

From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
Date: Tue, 14 Mar 1995 12:58:12 GMT
To: hfsig@tapr.org
Cc: nash@camail.ca.nmp.nokia.com
Message-ID: <199503141257.0AA23492@noknic.nokia.com>
Rendered body type: text/plain

Hello Andries

>Perhaps I will be forgiven for slipping in a question at this point
>that is slightly off the topic ? At the moment I am working on some
>data decoding software and one of the topics that came up was a 32
>-tone version of MFSK that Klingenfuss calls CIS Piccolo and
>apparently also is known as CROWD36. I can manage to extract the raw
>symbols, but of course the alphabet is the main problem. I was
>wondering : does anyone have any info on this, or could you point me
>in the direction of an information source ?

I presume that CIS Piccolo is pretty much the same as UK FCO Piccolo. It would seem reasonable to assume that the alphabet is the same as Cyrillic radio teletype using the NULL character as a 3rd shift. I think Mr Klingenfuss has described this alphabet in his book.

Thought for the day : I wonder if the UK Foreign Offices' old LA2028 Piccolos ended up in the CIS??? :-)

Regards

Adrian G4ZHZ

27. Re: [HFSIG:318] HF simulation

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 14 Mar 1995 08:22:45 PST8PDT
To: hfsig@tapr.org
Message-ID: <49190E670A6@frl.orst.edu>
Rendered body type: text/plain

Bill,

A good place to look for programs and some documentation on .wav files are in the SIMTEL archives (ftp to: oak.oakland.edu) and look in the windows/sounds directory for a collection of sound record/playback programs.

If you still have a question, I'll be happy to post the format.

73's

Johan

28. Re: [HFSIG:308] Thoughts on HF channel simulation

From: bm@lynx.ve3jf.ampr.org (Barry McLarnon VE3JF)
Date: Wed, 15 Mar 1995 04:23:40 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <m0rokcy-0002pQC@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

> Everything I read and hear says that HF channels are really hard to
> model and simulate realistically. But a repeatable simulation is
> really important to have if you want to do controlled tests and
> comparisons.
>
> Would it be possible to probe a real HF channel, record its response

> digitally, and then use that information in offline simulations that
> could be replayed as often as desired?

Definitely. I brought this up briefly with Johan at the TAPR meeting. About 10 years ago, when I was involved with HF stuff at my place of work, we had an HF channel simulator with this capability built under contract. The simulator used the TMS32010, shortly after that chip was first introduced. Separate from the simulator was a PRBS probe generator, clocked such that the main lobe would fit comfortably in the passband (~2 kHz) of an ssb transmitter. This was transmitted over various HF channels, and the baseband receiver output (with AGC off) recorded on an instrumentation tape recorder. Nowadays, of course, you'd use your sound card and store the received signal on your hard disk. The tape could then be played back into the simulator, where the signal was digitized and correlated in real time against a stored replica of the PRBS obtained from a back-to-back connection of the transmitter and receiver. The impulse response estimates from the correlator were then used to adjust the tap weights of the simulator in the main signal path. This is very convenient, since it allows you to reproduce the actual conditions on a real channel in a completely repeatable fashion, with no dependence on the modulation format of the system(s) under test.

I'll see if I can dig up some docs on the thing... I don't think there's anything proprietary about it - it was a "one of" development and was never made into a commercial product.

> Phil

Barry

--

Barry McLarnon VE3JF/VA3TCP
Ottawa Amateur Radio Club Packet Working Group
Email: bm@hydra.carleton.ca or bm@lynx.ve3jf.ampr.org

29. Re: [HFSIG:315] HF Channel simulator

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Wed, 15 Mar 1995 12:15:54 -0800
To: hfsig@tapr.org
Message-ID: <199503152015.MAA26223@unix.ka9q.ampr.org>
Rendered body type: text/plain

>2) Phil (I think) is interested in knowing the channel's actual transfer
>function. Knowing this, one may use adaptive techniques. This is

Not quite. I was mainly interested in finding a way to compare different modems (or the same modem with experimental modifications) in a repeatable and realistic way. Channel simulators based on analytical models are repeatable, but people are concerned about their realism. Actual on-air testing is certainly realistic, but it can be hard to repeat consistently.

So my idea was to capture an actual HF channel over a particular span of time (thus gaining realism) in such a way that it can be used repeatedly. If I could record the measured response of a real HF channel over some period of time, then a simulator could use it to corrupt and modify my test signal exactly as if it had been actually sent over the real channel. Even better, I would have the option of repeating exactly the same channel simulation sequence with a different modem and then comparing results on an even footing.

Of course, you wouldn't just measure an HF channel once, you'd want to collect a wide variety of runs on different bands, paths and ionospheric conditions.

Phil

30. Re: [HFSIG:322] Re: Thoughts on HF channel simulation

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Wed, 15 Mar 1995 17:27:20 -0800
To: hfsig@tapr.org
Message-ID: <199503160127.RAA26557@unix.ka9q.ampr.org>
Rendered body type: text/plain

>first introduced. Separate from the simulator was a PRBS probe
>generator, clocked such that the main lobe would fit comfortably in the
>passband (~2 kHz) of an ssb transmitter. This was transmitted over
>various HF channels, and the baseband receiver output (with AGC off)

>recorded on an instrumentation tape recorder. Nowadays, of course,

Bingo, this is exactly what I had in mind.

A possible alternative to the pseudo-random bit stream would be a collection of closely spaced CW carriers of known amplitude and phase across the passband of the SSB transceiver. You could then measure the amplitude and phase variations of each and use the in a FFT-based filter that simulates not only the HF channel, but the filters in the radios.

Mathematically, both approaches are of course pretty much equivalent, it's just a difference of implementation.

Phil

31. [HFSIG:322] RE: THOUGHTS ON HF CHANNEL S

From: wtb@partyline.org
Date: Wed, 15 Mar 95 22:09:11 -0600
To: hfsig@tapr.org
Message-ID: <9503152209.0V4DA00@partyline.org>
Rendered body type: text/plain

Isn't it true that those of us who are planning to do our DSP on the DSP93, a separate box from our computers, would want the channel simulation (recordings) as analog tapes???

Bill NOLIK

32. Re: [HFSIG:325] RE: THOUGHTS ON HF CHANNEL S

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 15 Mar 1995 21:44:22 -0800
To: hfsig@tapr.org
Cc: wtb@partyline.org
Message-ID: <199503160544.VAA27211@servo.qualcomm.com>
Rendered body type: text/plain

>Isn't it true that those of us who are planning to do our DSP on the
>DSP93, a separate box from our computers, would want the channel
>simulation (recordings) as analog tapes???

Not really. Who says I have to run my simulations on my DSP-93? Since they don't have to run in real time, I could do them much more easily on a general purpose PC.

Phil

33. Re: [HFSIG:324] Re: Thoughts on HF channel simulation

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Thu, 16 Mar 1995 08:55:43 PST8PDT
To: hfsig@tapr.org
Message-ID: <4C21EE46B0C@frl.orst.edu>
Rendered body type: text/plain

Phil,

>>first introduced. Separate from the simulator was a PRBS probe
>>generator, clocked such that the main lobe would fit comfortably in the
>>passband (~2 kHz) of an ssb transmitter. This was transmitted over
>>various HF channels, and the baseband receiver output (with AGC off)
>>recorded on an instrumentation tape recorder. Nowadays, of course,
>

I good idea I agree - this is actually what the Watterson group did, however, after characterising the channel, went ahead and built the simulation model. The Watterson paper is actually good reading about acquiring channel response.

I have read a few papers where "the channel" is implemented as a FIR filter. The FIR filter, in this case, having non-symmetric filter coefficients, thus able to produce most any amplitude/phase responses. The taps are then fed in real time from a stored data stream. This then gives "realism". I guess it would not be impossible to extract the coefficients for a given channel, given a procedure outline above.

>Bingo, this is exactly what I had in mind.

>

>A possible alternative to the pseudo-random bit stream would be a

>collection of closely spaced CW carriers of known amplitude and phase
>across the passband of the SSB transceiver. You could then measure the
>amplitude and phase variations of each and use the in a FFT-based
>filter that simulates not only the HF channel, but the filters in the
>radios.
>
>Mathematically, both approaches are of course pretty much equivalent,
>it's just a difference of implementation.
>
>Phil
>

Johan, KC7WW

34. Fano decoder uploaded to ftp.ucsd.edu

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Fri, 17 Mar 1995 10:35:18 -0800
To: tcp-group@ucsd.edu, hfsig@tapr.org
Message-ID: <199503171835.KAA00237@unix.ka9q.ampr.org>
Rendered body type: text/plain

My Fano (sequential) decoder package is now available as

ftp://ftp.ucsd.edu/hamradio/packet/tcpip/incoming/fano.tar.gz
ftp://ftp.ucsd.edu/hamradio/packet/tcpip/incoming/fano.zip

Three different rate 1/2 constraint length 32 convolutional codes are supported. The decoder operates on 8-bit soft decisions. A routine is provided to build the decoder metric table assuming gaussian noise at some estimated E_s/N_0 . By substituting different metric tables, the decoder can be adapted to run on other kinds of channels such as the binary symmetric channel (BSC), i.e., with hard decoding.

Using GCC 2.5.8 on a 66 Mhz 486DX2 in protected (32-bit) mode, the decoder runs at about 300,000 branches per second. This translates directly to 300,000 bits/sec on a "clean" channel. Noisy packets decode more slowly, as is characteristic for a sequential decoder. The performance curve is quite steep, as expected for a strong, long constraint length code. Simulation results show that at an E_b/N_0 of 4dB, only about 1.6% of the packets decode at half speed or slower. Below about 2.5-3dB, things start to fall apart as expected.

Although the encoder/decoder itself is in portable ANSI C, the test driver assumes a UNIX (specifically BSDI1.1) environment with curses/termcap. Also, the synthetic gaussian noise generator uses `log()` and `sqrt()` quite heavily. Some 386/486 math libraries don't use the native FPU instructions for these functions, so they can be quite slow.

My Viterbi (maximum likelihood) decoder is next.

--Phil

35. Re: [HFSIG:328] Fano decoder uploaded to ftp.ucsd.edu

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Sat, 18 Mar 1995 16:51:33 PST8PDT
To: hfsig@tapr.org
Message-ID: <4FA0F1155A0@frl.orst.edu>
Rendered body type: text/plain

Phil,

Thanks - that is great news. I'll be trying that out soon.

-Johan

36. Re: [HFSIG:327] Re: Thoughts on HF channel simulation

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sat, 18 Mar 1995 19:35:15 -0800
To: hfsig@tapr.org
Message-ID: <199503190335.TAA01816@unix.ka9q.ampr.org>
Rendered body type: text/plain

>I have read a few papers where "the channel" is implemented as a FIR filter.
>The FIR filter, in this case, having non-symmetric filter coefficients, thus
>able to produce most any amplitude/phase responses. The taps are then fed
>in real time from a stored data stream. This then gives "realism". I guess
>it would not be impossible to extract the coefficients for a given channel,

>given a procedure outline above.

Why not? Just send impulses down a real HF channel. What comes out of the receiver will be, by definition, the impulse response of the channel. These could be used in a time-varying FIR filter to later simulate the same channel as many times as you wish.

Phil

37. Viterbi decoder package released

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Sat, 18 Mar 1995 23:16:03 -0800
To: hfsig@tapr.org, tcp-group@ucsd.edu
Message-ID: <199503190716.XAA02636@unix.ka9q.ampr.org>
Rendered body type: text/plain

As promised, I have uploaded my Viterbi decoder package:

ftp://ftp.ucsd.edu/hamradio/packet/tcpip/incoming/viterbi.tar.gz
ftp://ftp.ucsd.edu/hamradio/packet/tcpip/incoming/viterbi.zip

This package implements a coder and soft decision Viterbi (maximum likelihood) decoder for the rate 1/2 K=7 NASA standard convolutional code. It includes a driver program for test purposes.

The decoder runs at about 45.5 kilobits/sec on a 66 MHz 486DX2 under BSDI1.1 and gcc 1.42. (Interestingly enough, it runs only about 40.8 kb/s under gcc 2.5.8, even with full optimization). This is about 6.6x slower than my Fano decoder on a "clean" channel. But unlike a Fano decoder, which bogs down on a noisy channel, the Viterbi decoder runs at a constant rate even when the signal is very weak (though not without errors, of course).

Phil

38. Re: [HFSIG:331] Viterbi decoder package released

From: Les Davies <ve6ahm@ve6ahm-1.ampr.ab.ca>
Date: Mon, 20 Mar 1995 01:52:02 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <199503200852.BAA00773@ve6ahm-1.ampr.ab.ca>
Rendered body type: text/plain

What can these packages be used for. Also what additional is required for them?

Les

39. Re: [HFSIG:332] Re: Viterbi decoder package released

From: Phil Karn <karn@qualcomm.com>
Date: Sun, 19 Mar 1995 16:27:10 -0800
To: hfsig@tapr.org
Message-ID: <199503200027.QAA12011@servo.qualcomm.com>
Rendered body type: text/plain

Both packages consist just of the encoder/decoder subroutines plus test drivers. By themselves they are primarily intended for experimentation and education. But you could also integrate them into a system if you are so inclined.

Phil

40. ACARS SOFTWARE WANTED

From: Simon Lewis <slewis@pacsat.demon.co.uk>
Date: Sun, 26 Mar 1995 21:48:19 GMT
To: hfsig@tapr.org
Message-ID: <22@pacsat.demon.co.uk>
Rendered body type: text/plain

Hi All

I'm in need of some ACARS decoding software for the aero data system. I know that this isnt an HF sig mode but any help from like minded data individuals would be appreciated.

--

```
+-----+
| 73 From: Simon Lewis GM4PLM - Helensburgh - Strathclyde - Scotland |
| Packet: GM4PLM @ GB7SAN.#78.GBR.EU Email: slewis@pacsat.demon.co.uk |
| AMSAT - UK Member 4282 **** SUPPORT YOUR LOCAL AMSAT GROUP **** |
+-----+
```

41. silence

From: wtb@partyline.org
Date: Sun, 26 Mar 95 16:59:27
To: hfsig@tapr.org
Message-ID: <9503261659.0NVAH00@partyline.org>
Rendered body type: text/plain

No info from SIG for over a week.. Is this normal or fault??
Bill NOLIK

42. HFSIG activities

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 27 Mar 1995 09:01:46 PST8PDT
To: HFSIG@tapr.org
Message-ID: <5CA409544FC@frl.orst.edu>
Rendered body type: text/plain

Hi All,

After my plea for help, I have not heard from anyone, so without anyone to help out, this is what probably will be happening for a while:

1) Don't despair - I have not lost interest. As soon as things settle down with my present time sink, there will be more time to spend on helping with the dissemination of ideas. With the way things are looking right now, this may be around September.

2) I am working with some on a few worthy projects - one of these are MFSK. It appears that this technology is far from dead and have taken a very interesting direction, for example: Viterbi decoding for both error-control and synchronisation. More on this at a later opportunity. The other is that I received a DSP-93 and will be porting a number of applications that I have running on the TI DSK to this platform. Initially, this will be the W9GR denoiser and the HF DSP modems for AMTOR/PACTOR.

Activity on HFSIG will be what you make of it, I'm sure there are lots of interesting ideas out there - use the opportunity.

73's

Johan, KC7WW

43. Re: [HFSIG:336] HFSIG activities

From: fperkins@onramp.net
Date: Mon, 27 Mar 1995 18:55:29 -0600
To: hfsig@tapr.org
Message-ID: <199503280055.SAA23494@ns.onramp.net>
Rendered body type: text/plain

Hi Johan,

Great to hear you have joined the DSP-93 development team!

73 Frank WB5IPM

44. [HFSIG:336] HFSIG ACTIVITIES

From: wtb@partyline.org
Date: Tue, 28 Mar 95 11:48:53
To: hfsig@tapr.org
Message-ID: <9503281148.0GLIH00@partyline.org>
Rendered body type: text/plain

Johan

Dont feel pressed, its only a hobby. I was merely checking, not pushing. You guys have finally pushed me into 486/50 with windows so I have weeks of studying to do. I'm also trying to get my hands on Phil's viterbi.zip thru ftpmail and so far have failed miserably. Good luck on your pursuits.

BILL NOLIK

45. Johan's Schedule

From: Greg Jones <gjones@tenet.edu>
Date: Tue, 28 Mar 1995 12:05:28 -0600 (CST)
To: hfsig@tapr.org (HF SIG mailing)
Message-ID: <199503281805.MAA25465@Alice-Thurman.tenet.edu>
Rendered body type: text/plain

Greeting HF SIG. Johan approached me before the TAPR annual meeting regarding the fact that he would be very busy preparing for his dissertation defense over the next 6 months. I can literally understand his position, as I am taking my candidacy exams next week.

Problem:

1. HF SIG will come to a complete halt over the next 6 months unless we find someone to step forward and at least provide the necessary facilitation to continue discussion.
2. Johan will return and be available to help, but just not to do all the talking during the period.
3. The group needs a scribe for the activity....Johan has been doing this as well..and I can understand why he might feel a little over worked :-)

So - who has just a little time in the next 6 months to help organize and report activity within the group....not to completely step into Johan's shoes...just to help keep the topic of discussion focused until Johan can return in full force and we can call him 'Dr. Forrer'.

We really need to find someone who will step forward and volunteer for either some or all of this, or the HF SIG will be in a reduction mode. I would hate to see that, based on the tremendous dialog and work that has been accomplished thus far.

If you think you can help - contact either Johan or myself.

Cheers - Greg

President -- Tucson Amateur Packet Radio Corp

TAPR Office (817) 383-0000 | Internet: gjones@tenet.edu

46. Re: [HFSIG:336] HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 29 Mar 1995 11:55 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <547C7D2B20A02883@chopin.ifj.edu.pl>
Rendered body type: text/plain

Let me say about what I am doing:

I am still at the stage of coding the parallel carrier modem into the DSPCARD4. The current code can send out the tones, with a reasonable initial phase set (so the Crest factor isn't too bad). The receiver's FFT works too. There are some provisions for shifting the receiver's sampling point for symbol synchronization.

Missing items:

1. Inter-symbol interference: how to send and receive symbols so they don't leak one into another across time and frequency. Yesterday I have worked out the problem (at last !) and I feel being myself on a straight way.
2. Bit encoding/decoding.
3. Symbol synchronization.
4. Tuning indication.

I believe I have good ideas to follow in every of these aspects so the whole project is only a matter of time...

By the way a little thing which came across my mind recently:

You remember that idea of putting same information at every carrier and then doing a majority vote at the receiver.
If a,b,c,... are the bits to be sent the coding pattern looks like this:

```
1 2 3 .... <- window number
1  a b c . . . .
2  . a b c . . . .
3  . . a b c . . .
4  . . . a b c . .
5  . . . . a b . .
6  . . . . . a b .
```

```

7   . . . . . a b .
8   . . . . . a b .
9   . . . . . a b .
10  . . . . . a b .
11  . . . . . a b .
12  . . . . . a b .
13  . . . . . a b .
14  . . . . . a b .
15  . . . . . a b .

```

^
carrier number

The intended feature of this scheme is time and frequency diversity. But this scheme has another interesting feature: being mistuned by one carrier makes little harm (your symbol timing shifts and you loose 2 data streams out of 15 but the message is still decodeable). I am not sure what happens if you get mistuned by a fraction of the carrier spacing...

Are there schemes which provide resistance again any (but not to large) mistune so you don't care much about tuning your receiver to these last Hertz's ?

I could think of one such scheme: our elementary symbol would be a constant rate sweep across the whole available band say 300..3000 Hz. At the receiver you listen to the sweep through several narrow filters (this is done with the help of FFT). Every filter gives you the same data stream but shifted in time. Now you have time and frequency diversity and reasonable insensitivity to mistuning your receiver (or possible Doppler shift if it doesn't change too fast with time).

Pawel

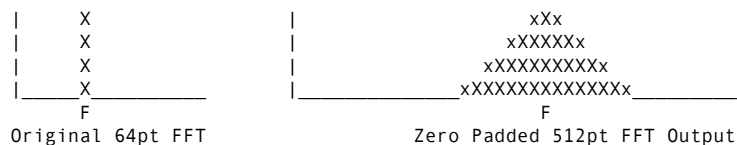
47. Re: [HFSIG:340] Re: HFSIG activities

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Wed, 29 Mar 95 09:37:41 nst
To: hfsig@tapr.org
Message-ID: <9502297964.AA796498681@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

Pawel,

One thing I have come across that may help with being off frequency when using an FFT is: zero padding.

That is you use an FFT several times larger than necessary, and fill the extra points with zeros.



The tones that were single "bars" in the original FFT, will now become "bumps" over the several new bars/bins around the frequency. If the tone was not on frequency in the original FFT, then one of the bins to one side of this frequency will have more energy in the new FFT. i.e. 8000Hz Sampling, 64point FFT = 125Hz/point. If zero pad the FFT to 512point = 15.625Hz/point which should be sufficient resolution for radio frequency error? And all you do is add up the power of the 8 groups of every (512/64=8) 8th point, and choose the point set that has the highest power as where the tones are centred.

Care must be taken in that there may be some distortion to the phases of the points, although my initial simulations showed this to be negligible.

There have also been suggestions to me, that since our tones are orthogonal, that instead of zero padding, you repeat the FFT input data to fill the larger FFT input buffer. I understand that in some tests this has worked well. But in my own simulations I have seen occasional strange results (human error?). I am also worried about what may happen if the radio is off frequency, causing the received tones to no longer be completely orthogonal, and the resulting phase discrepancies at the repeat boundaries to causes erroneous FFT output.

As well as comments on above by anyone, I am extremely interested in your methods for solving:

1. Inter-symbol interference
I am looking at shaping the signals using either $\sin()$ or $\sin()^2$, this reduced harmonics at the phase/frequency changes between symbols as the signal power drops to zero. But the shaping causes a large amount of energy in the adjacent FFT Bin. (Presently using $\sin()$ as higher centre to lobe ratio, and the further lobes are in the dirt anyway). Problem is shaping reduces the overall bit energy, reducing the SNR.
2. Bit encoding/decoding.
Are you planning an interleaved Golay code, or similiar? I've seen alot of coding software posted here.

I am uncertain if I correctly understand your diagram, showing the staggered bit output on easy tone successively. If this is done, then each bit is sent 16 times, won't some form of error correction coding be more efficient?
3. Symbol synchronization.
If you shape the signal as in 1 above, then tracking the average audio power gives you a clock.
4. Tuning indication.
If you use the Zero Padded FFT, then the Choice of point set gives a course tuning indication. A larger FFT would bring it in closer.

Paul Russell

48. HFSIG activities

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Wed, 29 Mar 1995 09:14:22 PST8PDT
To: hfsig@tapr.org
Message-ID: <5FA77D134D7@frl.orst.edu>
Rendered body type: text/plain

Hi All,

On a more positive angle - a short update:

Pawel - congratulations on your progress with the parallel modem. One interesting article that I have recently worked through, concerning symbol synchronisation for MFSK, uses the Viterbi algorithm. This, in essence is how they do it:

You need to compute a sliding FFT, at least over several instances per symbol, then feed the output of the FFT bins at each instance into a decoder trellis. Do this over several symbols. The output of the decoder now will tell you the best path, in maximum likelihood sense, through the trellis, i.e. at which instance you have the center of the symbol. The same trellis is then also used for data error-control. For a description of the paper:

"Improved HF modem using convolutional coding." by B. Honary, F. Zolghadr, M. Darnell, and M. Maundrell. In Electronics & Communications Engineering Volume 4(2) April 1992. (Thanks to Frode for the reference).

I also have been following Adrian's (G4ZHZ) work on MFSK. This is really outstanding - Adrian is busy dealing with some challenging issues. I'll let Adrian give us a overview of this when he is ready - better will encourage him to write an article about it! Just to fill you in a bit: One of these is how to deal with the distortion inherent in a numerical approach, i.e., local oscillator distortion, but also with dynamic range of 16-bit DSP for FFT's vs. a correlator (matched filter) approach. These things are easy to talk about but another matter to live with in your actual implementations. This is where one have to carefully weigh your choice in DSP chips - do you use the appropriate architecture, should you rather use a floating point device? Another idea is to use a marriage between analog and DSP techniques. On this matter, if anyone has some ideas on a good analog front-end processor that can produce high resolution I/Q outputs, I am very interested.

We are still looking for help with this SIG: As you can gather from the recent contributions, please consider helping out for the next few months.

Thanks,

Johan, KC7WW

49. DSP Floating Point Unit

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Wed, 29 Mar 95 14:13:13 nst
To: hfsig@tapr.org
Message-ID: <9502297965.AA796515241@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

FYI

I am uncertain if I mentioned it before, but the best deal I've seen on a Floating Point standalone DSP unit is the DSP Tools unit. It connects to a PC via a parallel port, or runs standalone. Has a 33MHz TMS320C31 (50MHz available), and FLASH ROM for code storage. Audio is via a TLC32040 AIC.

\$299.00 US (email Dave Jervis at DSP Tools for info:
davejervis@aol.com)

The floating point is easier than integer in that you don't have to worry about dynamic range of the signal (clipping etc.) during the stages of signal processing. But it costs more.

Query: How much is the DSP-93 and what processor does it use?

Paul Russell

50. Re: [HFSIG:341] Re: HFSIG activities

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 29 Mar 1995 12:26:30 -0800
To: paulr@hagar.udc.neweast.ca
Cc: hfsig@tapr.org
Message-ID: <199503292026.MAA23656@servo.qualcomm.com>
Rendered body type: text/plain

Paul,

Your zero-padded FFT idea is basically a rectangular sampling window that's shrunk to take up less than the whole time. Have you considered any of the other sampling windows? There are many to consider: Hamming, hanning, Kaiser, etc.

The right choice of sampling window is important in rejecting strong in-band interference in a m-ary orthogonal FSK scheme. Otherwise, the windowing artifacts would cause energy to spread over much of the spectrum after the FFT.

Phil

51. Re: [HFSIG:342] HFSIG activities

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 29 Mar 1995 12:32:00 -0800
To: hfsig@tapr.org
Message-ID: <199503292032.MAA23662@servo.qualcomm.com>
Rendered body type: text/plain

Johan,

It's amazing how many uses people have found for the Viterbi algorithm beyond straight FEC. Several textbooks discuss using it to do maximum likelihood decoding of signals with intersymbol interference. In the case of duobinary, a 3-level signalling scheme with deliberate, controlled intersymbol interference, a Viterbi decoder buys back *all* of the S/N loss that would otherwise result from going from 2 levels to 3.

Of course, the range of channels is somewhat limited by the Viterbi decoder's exponential complexity function.

Phil

52. Re: [HFSIG:341] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl
Date: Thu, 30 Mar 1995 15:15 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <3992D3FF80A03CC2@chopin.ifj.edu.pl>
Rendered body type: text/plain

Hello Paul,

>As well as comments on above by anyone, I am extremely interested in your
>methods for solving:


```
>
>1. Inter-symbol interference
>    I am looking at shaping the signals using either sin() or sin()^2,
```

After scratching my head a lot I decided to use $\sin()^2$...
The numbers I work with right now are:

Sampling: 9600 Hz
FFT size: 128 points -> 64 frequency bins between 0 and 4800 Hz
I take even bins for data transmission, odd ones are for tuning.
I use 15 carriers, the first one is #8 (600 Hz), the last one is 2700 Hz.
This fits into 300-3000 Hz band.

Window: $\sin(x)^2$ for $x=0..PI$, the total length of the window
is 128 samples (same as the FFT size). I use same window for transmit
and receive.

Spacing between symbols is 64 samples on I-part of the carrier
and same for the Q-part.
On both I- and Q-parts I use a BPSK coding: a 180 degrees change
in phase is a "0", no change is a "1". I- and Q- symbols are shifted
by 32 samples. This is the maximum bit packing you can do theoretically.

I'm going to try as well a carrier-on/carrier-off approach.

With $\sin()^2$ window I expect inter-symbol crosstalks of about 1/6
to a symbol adjacent in time or in frequency, -1/12 to symbols
"across diagonals". For a symbol on the carrier #12 the crosstalk
pattern looks like this:

```

    ---|-----|-----|----> time: |-----| = 64 samples

carrier #14:  -1/12  1/6  -1/12

carrier #12:   1/6    1    1/6

carrier #10:  -1/12  1/6  -1/12
```

For a worst case the total crosstalk from adjacent symbols
reaches 1 so it may overcome the symbol you are interested in.
I correct for crosstalk by subtracting adjacent symbols
according to the above pattern.

```
>    this reduced harmonics at the phase/frequency changes between symbols
>    as the signal power drops to zero. But the shaping causes a large amount
>    of energy in the adjacent FFT Bin. (Presently using sin() as higher
>    centre to lobe ratio, and the further lobes are in the dirt anyway).
>    Problem is shaping reduces the overall bit energy, reducing the SNR.
```

This is a very tricky thing: I would say that using rectangular window
(no shaping) makes your symbol's energy go to other frequency bins
(sounds the other way you say ?). Shaping makes the bit energy to concentrate
in the bin you want.

With rectangular window you don't see this energy being spread around
only because your other frequency bins are spaced just right.
If they were a bit off, you would notice...

>2. Bit encoding/decoding.

This point was the way I encode the bits into the carrier states (look above).

```
>    Are you planning an interleaved Golay code, or similiar? I've seen alot
>    of coding software posted here.
```

I foresee the following "simple" schemes:

0. no FEC at all (just to see how the modem alone works)
1. 11 data + 4 CRC bits FEC code capable of correcting single error
2. 15 bit Walsh functions (how many error can these correct ?)
3. "send same bit 15 times" with majority coding at the receiver
(cruel but can correct up to 7 errors).
4. with 24 carriers I might use a 12 data bits + 12 CRC bits Golay code
which can correct up to 3 errors.

1,2,3 and 4 would additionally involve spreading the bits in time
to gain certain resistance against error bursts.
For details about 1 and 3 look my previous posts... or I send them
to you if you haven't got them.

For easy start I will use use AX.25 framing and protocol: I take a HDLC frame and put the bits one by one (including the stuffing bits and the frame markers) onto the carriers.

> I am uncertain if I correctly understand your diagram, showing the
> staggered bit output on easy tone successively. If this is done, then
> each bit is sent 16 times, won't some form of error correction coding be
> more efficient?

This is right: every bit gets send 15 times but I am not saying this is an efficient FEC - this was only an example, just to present the idea of a signal which is little sensitive to a mis-tune at the receiver.

>3. Symbol synchronization.
> If you shape the signal as in 1 above, then tracking the average audio
> power gives you a clock.

Not so obvious I'm afraid... if you pack symbols as dense in time as I am trying to.

>4. Tuning indication.
> If you use the Zero Padded FFT, then the Choice of point set gives a
> coarse tuning indication. A larger FFT would bring it in closer.

Again not obvious when packing so dense in frequency as I am trying to do but I hope I would get the mis-tune indication by using the odd carrier information. Note, that for the moment I do not try to correct the mis-tune - I only want to indicate it so the operator corrects the receiver.

Progress (rather regress) report:

I have made up cross-talk correcting code yesterday - seems to work fine. Then I have put in the bit encode/decode (two BPSK's on I and Q shifted by half the symbol length). The result: constant bit patterns are passed well, but when I apply varying bit patterns I get some bits flipped...

Might be a symbol timing problem - the timing is fixed now - but adjusting the timing by hand didn't make much difference :-(
I hope it's a software bug somewhere and not a failure of a whole concept.

Pawel

53. Re: [HFSIG:344] Re: HFSIG activities

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Thu, 30 Mar 95 15:52:42 nst
To: hfsig@tapr.org
Message-ID: <9502307966.AA796607685@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

>>Your zero-padded FFT idea is basically a rectangular sampling window that's shrunk to take up less than the whole time. Have you considered any of the other sampling windows? There are many to consider: Hamming, hanning, Kaiser, etc.

The right choice of sampling window is important in rejecting strong in-band interference in a m-ary orthogonal FSK scheme. Otherwise, the windowing artifacts would cause energy to spread over much of the spectrum after the FFT.

Phil<<

I shape the Initial FFT Buffer before zero padding. I use either 1/2 a sine cycle ($\sin()$), or a raised cosine ($\sin()^2$). I also preshape the transmit symbols with either $\sin$ or $\sin^2$. The resultant receive symbols are thus shaped by either $\sin^2$ or $\sin^4$, as well as the noise being windowed. The Energy spread into adjacent FFT Bins is a problem.

I like the Preshaping of the Tx symbols at the symbol rate as it provides a clock to the receiver, and reduces Tx spikes/harmonics at the phase_changes/symbol_boundaries in DPSK. Although it also reduces your average Tx Energy per bit.

I'm doing simulations to analyse the Sliding FFT Window that Pawel is using, where the FFT input buffer is longer than the symbol.

I think a Preshaping with a $\sin()$ at the symbol rate (64pt), and a $\sin()^2$ windowing in the receiver that is twice the width of the symbol (128pt) is giving me my best results:

For DPSK, Each symbol 180degree shifted from previous: 64samples/symbol
 Receiver 128Pt FFT centred on Rx symbol.
 Best Combinations from simulations so Far:

Tx Shaping	Rx Windowing	128pt FFT Lobe Magnitudes (Only every 2nd point used) Centre: lobe 2 : lobe 4 : lobe 6				
None	Sin() ² 128pt	1	: 1/3	: 1/15	: 1/36	(Pawel's?)
Sin() 64pt	None	1	: 1/3	: 1/15	: 1/36	
Sin() 64pt	Sin() 64pt	1	: 1/2	: 0	: 0	
Sin() 64pt	Sin() ² 128Pt	1	: 1/2	: 0	: 0	

A 64 point FFT would also work, although the larger gives more frequency resolution. Test with a 512 point zero padded FFT gave the same ratios (at the corresponding bins) with more frequency resolution.

In simulations I have seen the windowing "artifacts" spread quite well across the band if the wrong window is chosen, or misaligned <grin>.

The purpose of the zero-padding is to give more frequency resolution by using a larger FFT, without increasing the symbol period. You do not get something for nothing as the "spikes" in the FFT spread out and it requires more memory and processing time, but it can help resolve where your carrier tones are (highest peaks) and thus indicate how far off frequency your radio may be.

Paul

54. Re: [HFSIG:346] Re: HFSIG activities

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
 Date: Thu, 30 Mar 95 16:01:34 nst
 To: hfsig@tapr.org
 Message-ID: <9502307966.AA796608226@ccsmtp.udc.neweast.ca>
 Rendered body type: text/plain

Pawel,

I have printed out all the messages I have received from hfsig re your design. I definitely need to re-read all of it slowly.

I'm having trouble understanding the advantage of the sliding window FFT, hopefully I'll find the light <g>.

One note: You mention you are planning on using tones from 600-2700Hz. I realize HF voice audio bands are supposed to be 300-3000Hz, but I have never seen a radio provide this. I would recommend dropping your upper tone, or shifting the whole down by 75Hz (525-2625). I have come across several commercial grade radios that only receive from ~375-2675Hz. I have even hit an older Harris RF230 that only passed 400-2500Hz. If the design is ment to operate on a wide range of radios, you may need to narrow your bandwidth requirements. Either lowering the sampling rate (8000Hz?) or using less tones may suffice. The equipment you are using may not have this limitation, this is just a warning.

Now off to study the printouts...

Paul

55. Re: [HFSIG:347] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl
 Date: Fri, 31 Mar 1995 17:57 GMT+1
 To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
 Message-ID: <195B304F80A0639F@chopin.ifj.edu.pl>
 Rendered body type: text/plain

Hello Paul (and others),

Tx Shaping	Rx Windowing	128pt FFT Lobe Magnitudes (Only every 2nd point used) Centre: lobe 2 : lobe 4 : lobe 6				
> None	> Sin() ² 128pt	> 1	> : 1/3	> : 1/15	> : 1/36	> (Pawel's?)
> Sin() 64pt	> None	> 1	> : 1/3	> : 1/15	> : 1/36	
> Sin() 64pt	> Sin() 64pt	> 1	> : 1/2	> : 0	> : 0	
> Sin() 64pt	> Sin() ² 128Pt	> 1	> : 1/2	> : 0	> : 0	

I preshape both at the Tx and at the Rx with 128 pt sin()².

>In simulations I have seen the windowing "artifacts" spread quite well across

>the band if the wrong window is chosen, or misaligned <grin>.

A bit different view at the Fourier Transform:

While doing the transform you effectively make several time-domain convolutions with these functions:

$\sin(k*2\pi*f*t)*\text{window}(t)$ and $\cos(k*2\pi*f*t)*\text{window}(t)$.

t is time, f is the base frequency of your FFT (sampling-rate/FFT-length).

You can regard the FFT output as a set of outputs from a series of FIR filters having the above shapes.

Now, if you want to separate your frequencies as much as possible, those FIR filters must be sharp. Not narrow, but sharp - that is the "walls" must be sharp and the out of band attenuation must be strong.

If you want a sharp FIR filter you must make it long. For low inter-carrier crosstalk the FFT length should be several times larger than the symbol time. For example try the FFT length of 256 with this window:

$\sin(\pi/64 * t)$ $t = -127.5 \dots 127.5$ (256 values to form a 256 point window)

You get 128 frequencies but you use every forth one to transmit symbols spaced at 64 points. Use the above window for both the receiver and the transmitter. You notice very low inter-carrier crosstalk at the price of having overshoot-type crosstalk inbetween successive symbols on same carrier.

I am now stuck with my ideas for phase-differential coding... getting the bits out while attempting to cancel the inter-symbol crosstalks and getting the symbol timing is too much. For the moment I am going to switch to carrier-on/carrier-off approach :-) and test few other ideas.

Pawel

56. Overlapped Symbol windowing (Was HFSIG Activities)

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Fri, 31 Mar 95 15:18:55 nst
To: hfsig@tapr.org
Message-ID: <9502317966.AA796692015@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

Pawel,

I'm Still having trouble understanding the advantages of the overlapped windowing on Tx and Rx instead of just windowing at the symbol rate. When there is a run of same phase symbols, the tone becomes continuous at the Tx, whereas when the phase is shifting the tone is "modulated" by the windowing. This leads to more energy/bit in streams of same phase symbols then in streams of alternating phase bits. Also it uses much more processing power and memory.

I have done up simulations, and anyone can have the file/printout. Just email the request and I'll email/fax back the results (simulation plots). The file shows the effects of phase modulating with 128pt $\sin()^2$ shaping on 64pt overlapped bits for a single channel/tone.

Word Document 160K
Postscript File 500K
Or I'll Fax It

The result of adding 128pt $\sin()^2$ shaped symbols, overlapped at 64 pts, of alternating phases (± 90 degrees = 180degree shifted) is the same as a 64pt $\sin()$ shaped signal. (Unless there is no phase shift which gives continuous tone). This is what my simulations produced. Is this correct, or Do I have a flaw in my data?

Also, By shaping the symbols so that they are timewise independent, we may be able to use multiple phases when the channel is better, thus doubling (4phases=2bits) or tripling (8phases=3bits) or data rate? Although the protocol would have to detect and adjust for this?

Paul Russell
paulr@neweast.ca

If I don't figure this out soon, you are all either going to shoot me or start

ignoring me :-)

April 1995

Original source file: hfsig.9504

Source: <https://archive.org/details/tapr-1997>

Recovered from: https://archive.org/download/TAPR-HF_1995-04/TAPR-HF_1995-04.txt

1. Parallel carrier modem

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>
Date: Sat, 1 Apr 95 13:52:21 MET
To: hfsig@tapr.org
Message-ID: <357.jalocho@home.ifj.edu.pl>
Rendered body type: text/plain

Hello everybody,

At last I can report a success: I became less ambitious :-)
and let my parallel carrier modem do only BPSK on every carrier.
The whole thing began to work ! At least it passes right varying
bit patterns and the symbol clock converges right.

The modem runs now at 9600 Hz sampling, 128 point FFT with $\sin()^2$
window at Tx and Rx. FFT steps are 32 samples with +/- 1 adjustments
for the receiver to synchronize it to the symbol clock.
Data carriers are spaced 150 Hz apart. Each carrier is BPSK modulated
with symbols (bits) spaced by 64 samples which makes 150 bits/sec.
There are 15 carriers: the first at 600 Hz and the last at 2700 Hz.
Total data rate is 2250 bps. No FEC coding yet.
Is this worth anything ?

Most of the above parameters can be easily adjusted: I can move
to 256 point FFT and 31 carriers immediately - more carriers very
likely need some tricks as I will run into my RAM limits.

The symbol timing and data carrier detection logic work somehow
although they need some tuning still. I have two different ideas
about how to make a tuning indicator - one of them must work :-).
In a short time I should have something useable for on-air operation.

I suspect there are still minor problems with inter-symbol crosstalks...
The way I attempt to cancel them is not the best one.
I could cure the problem by spacing the carriers by more Hz
- use every third carrier not every second.
The price: less carriers -> lower total bit rate.

Paul and others are certainly right that my carrier configuration
is too tight for an SSB transceiver. But moving to 8000 Hz sampling
is not a problem - just one line in the source code.
Or I just use less carriers, although 15 or 31 carriers
is a good number for my primitive FEC code.
24 is good for the Golay code.

At same time I am thinking about the non-coherent modulation,
and I can't see a good solution for the problem of finding
the threshold for 0/1 decisions. With BPSK the threshold is zero
regardless of carrier's amplitude, for carrier on/off modulation
the threshold can be potentially different for every carrier.

Pawel

2. Re: [HFSIG:350] Overlapped Symbol windowing (Was HFSIG Activities)

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>
Date: Sat, 1 Apr 95 15:08:35 MET
To: hfsig <hfsig@tapr.org>
Message-ID: <361.jalocho@home.ifj.edu.pl>
Rendered body type: text/plain

Hello Paul,

> The result of adding 128pt $\sin()^2$ shaped symbols, overlapped at 64 pts, of
> alternating phases (+/-90degrees = 180degree shifted) is the same as a 64pt
> $\sin()$ shaped signal. (Unless there is no phase shift which gives continuous
> tone). This is what my simulations produced. Is this correct, or Do I have a
> flaw in my data?

I think what you find is right.
I try to explain the point by the following (a bit naive) comparison:

shaping #1: 128 pt, $\sin()^2$

=====

bit pattern	waveform	bandwidth
0101010101	$\text{carrier} * \sin(2 * \pi * (1/128) * t)$	$2 * (1/128)$
1111111111 or 0000000000	pure carrier	0

shaping #2: 64 pt, sin()
=====

bit pattern	waveform	bandwidth
0101010101	$\text{carrier} * \sin(2 * \pi * (1/128) * t)$	$2 * (1/128)$
1111111111 or 0000000000	$\text{carrier} * \text{abs}(\sin(2 * \pi * (1/128) * t))$	infinite ! (or at least very large)

As you see the 64pt, sin() shaping generates a very wide signal for a constant bit pattern thus you cannot say, that your energy goes into the band assigned for that carrier. The harmonics (due to "abs" function) will affect other carriers.

> Also, By shaping the symbols so that they are timewise independent, we may be
> able to use multiple phases when the channel is better, thus doubling
> (4phases=2bits) or tripling (8phases=3bits) or data rate? Although the protocol
> would have to detect and adjust for this?

I agree the 64pt sin() shaping makes the symbols time-independent, but at the price of the bandwidth (thus frequency dependence). Here is the compromise to be taken...

Pawel

3. Re: [HFSIG:351] Parallel carrier modem

From: Phil Karn <karn@qualcomm.com>
Date: Sat, 1 Apr 1995 13:04:31 -0800
To: hfsig@tapr.org
Message-ID: <199504012104.NAA28192@servo.qualcomm.com>
Rendered body type: text/plain

>At same time I am thinking about the non-coherent modulation,
>and I can't see a good solution for the problem of finding
>the threshold for 0/1 decisions. With BPSK the threshold is zero
>regardless of carrier's amplitude, for carrier on/off modulation
>the threshold can be potentially different for every carrier.

This is only a problem with on-off keying, it is not inherent to non-coherent demodulation. Ordinary FSK is noncoherent, for example. So is m-ary FSK. You just look for the filter output with the greatest amplitude.

Phil

4. [HFSIG:353] Re: Parallel carrier modem

From: "Pawel Jalocha" <jalocha@home.ifj.edu.pl>
Date: Sun, 2 Apr 95 02:16:07 MET
To: hfsig <hfsig@tapr.org>
Message-ID: <364.jalocha@home.ifj.edu.pl>
Rendered body type: text/plain

Phil wrote:

> This is only a problem with on-off keying, it is not inherent to
> non-coherent demodulation. Ordinary FSK is noncoherent, for example.
> So is m-ary FSK. You just look for the filter output with the greatest
> amplitude.

Right, but only if you take the assumption that your passband is flat. I can see several reasons for the passband to be non-flat within at least 6dB thus a "take the highest" decision may not be always optimal. Ideally, you would monitor the levels on every carrier and compute individual thresholds. On the other hand I don't like the m-ary FSK as if somebody puts on a dead carrier the whole trasmission is trashed.

Which way would you suggest for Walsh function type signaling where you get many carriers turned on at a time ?

Pawel

5. Re: [HFSIG:354] Re: Parallel carrier modem

From: Phil Karn <karn@qualcomm.com>
Date: Sun, 2 Apr 1995 23:09:25 -0700
To: hfsig@tapr.org
Message-ID: <199504030609.XAA01986@servo.qualcomm.com>
Rendered body type: text/plain

>Right, but only if you take the assumption that your passband is flat.

So correct it if it's not flat. This is not a hard problem.

>On the other hand I don't like the m-ary FSK as if somebody
>puts on a dead carrier the whole trasmission is trashed.

Not at all. I have been thinking about this exact problem, and it turns out that there's a simple solution.

First, scramble your data. That ensures that the probability of two consecutive symbols (frequencies) being the same is $1/N$, where N is the number of distinct symbols/frequencies.

Second, the demod "copies behind". That is, it keeps a history of several symbols worth of FFTs. If it sees a consistent strong signal in one frequency bin, it attributes it to a strong in-band jammer and ignores that channel until it disappears. I.e., it picks the next-strongest channel as its output.

No real problem if the desired signal happens to hit on the jammer's frequency since this will only happen with probability $1/N$. Just pick a FEC code strong enough to correct this many errors.

My thinking here is that with this scheme for suppressing interference, the Walsh function layer may turn out to be unnecessary.

Proper choice of a window for the FFT will be important, though. Rejecting a very strong inband CW interferer requires a window with low sidelobes, but windows with this property also broaden the mainlobe. This may mean more bandwidth than would otherwise be required for $1/T$ orthogonality.

Phil

6. Re: [HFSIG:355] Re: Parallel carrier modem

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 5 Apr 1995 14:27 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <E9DDF4DC00A038C3@chopin.ifj.edu.pl>
Rendered body type: text/plain

>Second, the demod "copies behind". That is, it keeps a history of
>several symbols worth of FFTs. If it sees a consistent strong signal
>in one frequency bin, it attributes it to a strong in-band jammer and
>ignores that channel until it disappears. I.e., it picks the
>next-strongest channel as its output.

OK, but the algorithm should be enchanced so it can pick up more than one jammer :-)

Walsh function approach looks efficiant to me in this respect: if I have 128 or 256 carriers I can remove say 4 strongest ones and still be able to decode the symbol. An alternative approach would be to compute the average of all carriers and then remove carriers above 3 times the average or so.

Pawel

7. Re: [HFSIG:351] Parallel carrier modem

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 5 Apr 1995 15:03 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <EEEFB45EC0A038C3@chopin.ifj.edu.pl>
Rendered body type: text/plain

It's time to say something about my progress with the parallel-carrier BPSK modem:

The design of using minimal carrier spacing turned out to be rather unstable. It did work but crosstalks across time and frequency made the symbol-sync not very stable in low S/N. My attempts to cancel the crosstalk made the symbol-sync even less stable...

So I decided to use every third carrier, not every second one. In such configuration, with $\sin(x)^2$ window the crosstalk across carriers is zero. There is still left a crosstalk of 1/6 across symbols on same carrier, but the modem was behaving far better - I can say it was "useable".

So now the modem does 8000 Hz sampling, 128 point FFT. Data carriers are spaced at $3 \cdot (8000/128) = 187.5$ Hz each carrier is modulated with BPSK at 125 bps (64 points between symbols). I use 12 carriers which fit into the "standard" audio passband.

I am playing with two shaping windows (Tx and Rx use identical windows):

1. $\sin(x)^2$, $x=0..pi$
=> freq. crosstalk = 0, time crosstalk=1/6
2. $3/4 \cdot \sin(x) - 1/4 \cdot \sin(3x)$, $x=0..pi$
=> freq. crosstalk = 1/20, time crosstalk=1.7/20

The second window works a bit better. I don't try any more canceling the crosstalks - I just leave it as it is.

I tested the modem only in loopback mode. To check the behaviour in low S/N I was adding some noise from my HF receiver phone-jack. The bit-sync and DCD is stable up to very low S/N when you can hardly hear the tones in the noise. I have made up a KISS interface to the modem so now it is a useable KISS TNC. Somebody willing to play with it ? Note that the modem works on a DSPCARD4.

I still see a little problem with symbol-sync: it's a bit slow and needs TxDelay time of about 50 (500 ms).

I tried to come back to QPSK with "every third carrier design": the bit-sync locks up and the data looks almost OK but there are single flipped bit.

I plan to double the number of carriers to 24 and then add a Golay FEC code with interleave to see how it improves the copy. I only need to write Golay routines which don't take up too much space for decoding tables, so it all fits into my DSPCARD4 RAM.

Pawel

8. Re: [HFSIG:356] Re: Parallel carrier modem

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 5 Apr 1995 10:57:28 -0700
To: hfsig@tapr.org
Message-ID: <199504051757.KAA06366@servo.qualcomm.com>
Rendered body type: text/plain

The algorithm is easily extended to handle several strong jammers, limited by two things: the FFT window sidelobes and the strength of the FEC.

Phil

9. Proposal for a picollo-type modulation scheme

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>
Date: Mon, 10 Apr 95 17:37:20 MET
To: hfsig@tapr.org
Message-ID: <310.jalocho@home.ifj.edu.pl>
Rendered body type: text/plain

Last weekend I was doing some calculations about the performance of various modulations and the picollo-type modulation came out to be a clear winner. My thoughts drifted towards m-ary FSK and I have here a proposition for a picollo system:

I am bounded to my DSP system so just to fix some numbers

and to show the relations between them here is the characteristic of the FFT which is the base for modulator/demodulator:

Sampling rate: 8000 Hz.
FFT size: 256 samples -> 128 frequency bins.
FFT window: $\sin(x)^2$ for $x = 0..pi$.
FFT would be done in overlapping steps of 64 samples
Symbol length = 256 samples
Inter-symbol spacing = 128 samples (crosstalk = 1/6).
Symbol rate = $8000/128 = 62.5$ Hz
Inter-carrier spacing = 2 freq. bins (crosstalk = 1/6) = $2*(8000/256) = 62.5$ Hz
Carriers used: 32 -> total bandwidth = $32*2*(8000/256) = 2000$ Hz.
Bits per symbol: 5 -> raw bit rate = $5*62.5 = 312.5$ bps.

Tuning is a real problem with these HF modems.
If you get a radio with 1Hz resolution and stability this may seem not to be a problem but at least with my rig which has the tuning resolution of 10Hz and the stability of +/- 50-100 Hz this is a problem.
Even making a tuning indicator which tells you the de-tune within a single carrier is not enough in most cases.

To let the receiver auto-tune to the incoming signal at the beginning of a transmission we send a special signal: a ramp-up followed by a ramp-down, like this: (an example for 8 carriers)

```
Carrier
|
7      *
6      * *
5      * *
4      * *
3      * *
2      * *
1      * *
0      * *
01234567 -----> symbol
```

For my FFT setup with 32 carriers this would take 63 symbols which is a bit more than one second.

The receiver would hunt for ramp-up and ramp-down at same time and if a coincidence of both is found within the expected time the receiver is able to determine the band edges.

Note that the Rx doesn't need to find a complete ramp to determine the edges, it's enough if it finds one piece of the ramp-up and another piece of the ramp-down.
Because the ramps span the whole band there is minimal chance that narrowband interferences would prevent the receiver from detecting and measuring them.

Not only the band edges can be determined but the symbol timing can be found as well. Thus after detecting the sync. signal we are tuned and synchronized.

We can use the sync. for one more purpose:
to fix the FEC block timing. Our FEC coding needs the data to be sent in blocks thus the receiver should know where are the block's edges.
The moment when the ramp changes the direction can be considered as the mark point: 32 symbols later, when the ramp-down is complete, the first FEC block would start.

The sync. signal could be repeated several times if needed.
In this case a nice size for the FEC block would be $2*32 = 64$ symbols (this makes 320 bits or 40 bytes) - then whatever sync. we acquire, we will get synchronized to the FEC block timing.

Infact, this ramp-up, ramp-down series can be the base for the "scrambling" algorithm. That is if we send all zeros (NULL characters), the signal follows this series.

As you see from my FFT characteristics I "oversample" the signal by two both in time and in frequency. I think it should be possible to interpolate the points if we found that we are de-tuned by half or a quarter of a carrier. Same about the symbol timing shift. This should save the trouble with making the sliding FFT with time-adjust capability.

The way to hunt for ramps: I am going to try summing up energies along diagonals, like this:

```

Carrier
|
7          abc
-          abc
6          abc
-          abc
5          abc
-          abc
4          abc
-          abc
3          abc
-          abc
2          abc
-          abc
1          abc
-          abc
0          abc
0-1-2-3-4-5-6-7 -----> symbol

```

"-" = the "oversampled" points.
 "a" = points to be summed into A
 "b" = points to be summed into B
 "c" = points to be summed into C

If I find $B > \text{noise}$ and $B > A$ and $B > C$ I assume the ramp-up is around B and I get it's fine position T_u but interpolating with A and C.

Similar way for ramp-down: I find it's fine time T_d .

If I see $T_d - T_u = 8$ symbols (+/-tune-error) I compute:

```

Tu := Tu + 8 Symbols
Tune_correction = (Tu - Td)/2
Symbol_time = (Tu + Td)/2

```

Little associated ideas:

- Use Gray code - in a rather probable case that a particular symbol "moves" to the adjacent carrier due to receiver mistune or instability or the Doppler shift or anything, this will result in corruption on one bit only.
- Band equalization: I think it is enough to adjust the gains for every frequency bin such that the RMS of the outcoming signal is same for all bins. This should minimize destructive effects coming from dead carriers and CWs.

So what do others think of the whole idea ?

I am going to write some code for detecting the sync. to see how it behaves with low S/N.

Pawel

10. Re: [HFSIG:359] Proposal for a picollo-type modulation scheme

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
 Date: Mon, 10 Apr 1995 18:36:48 +0200 (MESZ)
 To: hfsig@tapr.org
 Message-ID: <199504101636.SAA21519@hades>
 Rendered body type: text/plain

Pawel wrote:

```

>
> Last weekend I was doing some calculations about the performance
> of various modulations and the picollo-type modulation came out
> to be a clear winner. My thoughts drifted towards m-ary FSK
> and I have here a proposition for a picollo system:

```

I is clear that m-ary FSK is better than i.e. OFDM, but you trade bandwidth for S/N. I believe that a modulation scheme with parallel orthogonal carriers (OFDM) and a decent coding will be better in overall performance (taking into account that the symbol time is limited to about 10ms due to multipath and thus to achieve high bitrate you have to use many many carriers in MFSK).

```

>
> I am bounded to my DSP system so just to fix some numbers
> and to show the relations between them here is the characteristic

```

```

> of the FFT which is the base for modulator/demodulator:
>
> Sampling rate: 8000 Hz.
> FFT size: 256 samples -> 128 frequency bins.
> FFT window: sin(x)^2 for x = 0..pi.
> FFT would be done in overlapping steps of 64 samples
> Symbol length = 256 samples
> Inter-symbol spacing = 128 samples (crosstalk = 1/6).
> Symbol rate = 8000/128 = 62.5 Hz
> Inter-carrier spacing = 2 freq. bins (crosstalk = 1/6) = 2*(8000/256) = 62.5 Hz
> Carriers used: 32 -> total bandwidth = 32*2*(8000/256) = 2000 Hz.
> Bits per symbol: 5 -> raw bit rate = 5*62.5 = 312.5 bps.

```

With 32 carriers and 4QPSK on each of them you would have 4000bps. Of course the S/N would degrade the performance of such a modulation scheme, I think you can win back a lot with coding. But to occupy a whole SSB channel for 312.5bps uncoded data is too inefficient.

The next problem is the robustness of such a system. I believe if one or more frequency bins fade out because of frequency selective fading you are in trouble. You would at least have to do some kind of interleaving to overcome these effects.

In the OFDM case we have several possible solutions. If we know the channel capacity of each bin or carrier there is a solution (water pouring) how to distribute the transmit power over the carriers. This implies that there is some back channel from the demodulator at the RX side to the TX side. This is possible in an ARQ system. If there is no such channel it is still possible to achieve a reasonable throughput by interleaving the information on the carriers, so that each virtual "carrier" carries the same mean information.

The third possibility is the application of broadband signals, OCDM (= orthogonal code division multiplex). I've read through an interesting article on this topic recently (J. Lindner, "Channel Coding and Modulation for Transmission over Multipath Channels", to be published in no. 3 AEUE, May 1995).

Alexander

```

--
*-----*
|      Alexander F. Kurpiers      |
|      Institut fuer Uebertragungstechnik      |
|      Merckstrasse 25      |
|      D-64283 Darmstadt (Germany)      |
|      Voice: +49-6151-162369      |
|      Fax  : +49-6151-165545      |
|      EMail: kurpiers@zeus.uet.e-technik.th-darmstadt.de      |
|      Hamradio: dl8aau@db0kg.#nds.deu.eu      |
*-----*

```

11. Re: [HFSIG:360] Re: Proposal for a picollo-type modulation

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Mon, 10 Apr 1995 15:38:30 PST8PDT
To: hfsig@tapr.org
Message-ID: <71FC914213E@frl.orst.edu>
Rendered body type: text/plain

Hi,

My \$0.02:

Alexander wrote:

```

]
]Pawel wrote:
]>
]> Last weekend I was doing some calculations about the performance
]> of various modulations and the picollo-type modulation came out
]> to be a clear winner. My thoughts drifted towards m-ary FSK
]> and I have here a proposition for a picollo system:
]
]I is clear that m-ary FSK is better than i.e. OFDM, but you trade
]bandwidth for S/N. I believe that a modulation scheme with parallel
]orthogonal carriers (OFDM) and a decent coding will be better in
]overall performance (taking into account that the symbol time is
]limited to about 10ms due to multipath and thus to achieve high
]bitrate you have to use many many carriers in MFSK).
]

```

```

]>
]> I am bounded to my DSP system so just to fix some numbers
]> and to show the relations between them here is the characteristic
]> of the FFT which is the base for modulator/demodulator:
]>
]> Sampling rate: 8000 Hz.
]> FFT size: 256 samples -> 128 frequency bins.
]> FFT window: sin(x)^2 for x = 0..pi.
]> FFT would be done in overlapping steps of 64 samples
]> Symbol length = 256 samples
]> Inter-symbol spacing = 128 samples (crosstalk = 1/6).
]> Symbol rate = 8000/128 = 62.5 Hz
]> Inter-carrier spacing = 2 freq. bins (crosstalk = 1/6) = 2*(8000/256) =
62.5 Hz
]> Carriers used: 32 -> total bandwidth = 32*2*(8000/256) = 2000 Hz.
]> Bits per symbol: 5 -> raw bit rate = 5*62.5 = 375 bps.
]
]With 32 carriers and 4QPSK on each of them you would have 4000bps. Of
]course the S/N would degrade the performance of such a modulation scheme,
]I think you can win back a lot with coding. But to occupy a whole SSB
]channel for 375bps uncoded data is too inefficient.
]
]The next problem is the robustness of such a system. I believe if one or
]more frequency bins fade out because of frequency selective fading you are
]in trouble. You would at least have to do some kind of interleaving to
]overcome these effects.
]

```

Agreed: If we want to take advantage of reasonable conditions, especially for networking, OFDM with appropriate coding would be great.

On the other hand, if we are interested in HF communications, no matter how adverse conditions may be, MFSK probably would be a better choice.

Implementation of a reasonable number of frequency slots, either for OFDM or MFSK may require a careful tradeoff in your algorithmic approach. I suspect there are further considerations when choosing between a correlator approach, as typically used for MFSK, or the FFT approach as typically used by the OFDM. It is my opinion that the simplicity of a well-designed correlator, i.e., taking advantage of decimated pre-processing, integrate/dump, and precision combiners, is superior in performance to an FFT approach - judged merely by roundoff errors and possible dynamic range. The catch is that such a correlator takes a lot more code and cannot compete with the efficiency of the FFT approach when it comes to a multitude of channels.

My guess, at least when considering our present fixed point DSP's is that MFSK will be limited to fewer channels - OFDM will handle more channels, however, will not be as robust.

I think that there is merit for both approaches and would encourage experimentation in either - who knows, perhaps there is someone out there that has a solution to increasing the dynamic range of the FFT approach, or someone that have a solution to implementing efficient correlators over a multitude of channels. In my modest opinion, those are our present challenges.

```

]In the OFDM case we have several possible solutions. If we know the channel
]capacity of each bin or carrier there is a solution (water pouring) how
]to distribute the transmit power over the carriers. This implies that there
]is some back channel from the demodulator at the RX side to the TX side.
]This
]is possible in an ARQ system. If there is no such channel it is still
]possible to achieve a reasonable throughput by interleaving the information
]on the carriers, so that each virtual "carrier" carries the same mean
]information.
]

```

```

]The third possibility is the application of broadband signals, OCDM
](= orthogonal code division multiplex). I've read through an interesting
]article on this topic recently (J. Lindner, "Channel Coding and Modulation
]for Transmission over Multipath Channels", to be published in
]no. 3 AEUE, May 1995).
]

```

Alexander, perhaps you could give us a summary of what this involves - thanks.

73's, Johan KC7WW

12. Re: Proposal for a picollo-type modulation

From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
Date: Tue, 11 Apr 1995 10:07:37 +0100
To: hfsig@tapr.org
Cc: nash@camail.ca.nmp.nokia.com
Message-ID: <199504110908.MAA17541@noknic.nokia.com>
Rendered body type: text/plain

Alexander wrote:

>I is clear that m-ary FSK is better than i.e. OFDM, but you trade
>bandwidth for S/N. I believe that a modulation scheme with parallel
>orthogonal carriers (OFDM) and a decent coding will be better in
>overall performance (taking into account that the symbol time is
>limited to about 10ms due to multipath and thus to achieve high
>bitrate you have to use many many carriers in MFSK).

My opinion is that MFSK is the optimum choice for single HF channel working, low-speed data. OFDM is for higher speed working. Both systems need to be developed. Perhaps OFDM could be made to degenerate gracefully towards MFSK in worsening channel conditions. If we think of OFDM as a number of multiplexed MFSK channels, then these channels could be made to drop out so that the same transmitter power becomes concentrated across a smaller bandwidth thus gaining SNR.

Johan wrote:

> Implementation of a resonable number of frequency slots, either
> for OFDM or MFSK may require a careful tradeoff in your algorithmic
> approach. I suspect there are further considerations when choosing
> between a correlator approach, as typically used for MFSK, or the
> FFT approach as typically used by the OFDM. It is my opinion
> that the simplicity of a well-designed correlator, i.e., taking
> advantage of decimated pre-processing, integrate/dump, and precision
> combiners, is superior in performance to an FFT approach - judged
> merely by roundoff errors and possible dynamic range. The catch is
> that such a correlator takes a lot more code and cannot compete with
> the efficiency of the FFT approach when it comes to a multitude of
> channels.

I have substituted a quadrature matched correlator type demodulator for the FFT-based demodulator in my ADSP2101 MFSK modem and so far have found that the Block Floating Point FFT with multiple stages of decimation is better. I do the post-integration arithmetic in full floating point. I am not yet clear as to why the correlator seems worse. The peak frequency bin is currently selected by scanning the array of magnitude squared results, comparing first the exponents of $M(n)$, M_{peak} , and setting $M_{peak}=M(n)$ if $M(n)$'s exponent is greater. If the exponents are the same, the mantissas are compared in a similar way. Perhaps this is not the best way of doing it. Any suggestions?

These are the figures for the matched correlator demodulator:

Input sample rate: 7680Hz
Input filter: 45 tap FIR $f_c=660\text{Hz}$
Decimation: 1/3
Correlator sample rate: 2560Hz
sin/cos lookup table: 256 entries (FTF=10Hz)
frame buffer size: 128 (50ms)
M correlators (M is typically 12) executed at 320Hz rate (16 x per 50ms symbol)

The point is, MFSK based on FFT is not bad and is very efficient compared to the matched correlator approach. It is also potentially upgradable in real-time to OFDM. The crucial issues for the MFSK DSP system are: aliasing and roundoff noise caused by demodulation to I/Q and decimation, FFT dynamic range and tuning control.

The idea of trying the correlators at signalling frequencies, was to get rid of the need for heavy decimation and quadrature demodulation ahead of the FFT, considerably simplifying the processing and hopefully reducing aliasing noise. So far, no advantage in doing this has been observed. Maybe I should try substituting the FFT for a bank of floating point matched correlator detectors which would offer more dynamic range compared to the CBFP FFT.

Perhaps we can try Pawel's suggestions of windowing and extended FFT resolution for MFSK systems thus departing from traditional MFSK processing to viewing MFSK as a pure spectral analysis problem.

Regards
Adrian G4ZHZ

13. Re: [HFSIG:360] Re: Proposal for a picollo-type modulation scheme

From: JALOCHA@chopin.ifj.edu.pl
Date: Tue, 11 Apr 1995 11:33 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <889902D900A0869A@chopin.ifj.edu.pl>
Rendered body type: text/plain

>With 32 carriers and 4QPSK on each of them you would have 4000bps.

This was exactly the number I got fascinated with.
When it came to the actual implementation, I found out I have to space carriers a bit wider to reduce crosstalk between them so I came down to 3000 bps. Then, the DQPSK did not work... only DBPSK did (but now I have an idea why and I will come back to QPSK) so I came down to 1500 bps - still not too bad.
Timo and Joni from Finland did some tests of this modem and actual performance was real bad. I suspect the tuning is a serious issue here. Their signal levels were S3..5 at the distance of about 15 km.

>Of course the S/N would degrade the performance of such a modulation scheme,
>I think you can win back a lot with coding. But to occupy a whole SSB
>channel for 375bps uncoded data is too inefficient.

The argument to support such inefficiency is that you can tolerate interference from other users whose signals come into your band.

>The next problem is the robustness of such a system. I believe if one or
>more frequency bins fade out because of frequency selective fading you are
>in trouble. You would at least have to do some kind of interleaving to
>overcome these effects.

This issue was discussed here last week and we count on our forward error correction code to restore symbols missed due to selective fading or narrowbanded interferences.

>The third possibility is the application of broadband signals, OCDM
>(= orthogonal code division multiplex). I've read through an interesting
>article on this topic recently (J. Lindner, "Channel Coding and Modulation
>for Transmission over Multipath Channels", to be published in
>no. 3 AEUE, May 1995).

Would be very nice if you could say something more on the ideas expressed in this article.

Yesterday I have made up a band equalization and a ramp-hunting code as I described it yesterday (no precise ramp measurement yet, just the coincidence of ramp-up with ramp-down).
I wanted to see how reliable the detection is in the presence of wideband and narrowbanded noise. I found my ears are just a bit better :-)
but basically the sync. could still be detected when I could hardly hear it.
I should make some more solid signal/noise measurements...

Pawel

14. Re: [HFSIG:362] Re: Proposal for a picollo-type modulation

From: JALOCHA@chopin.ifj.edu.pl
Date: Tue, 11 Apr 1995 14:36 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <A213D25340A0869A@chopin.ifj.edu.pl>
Rendered body type: text/plain

Adrian wrote:

>My opinion is that MFSK is the optimum choice for single HF channel working,
>low-speed data. OFDM is for higher speed working. Both systems need to be
>developed. Perhaps OFDM could be made to degenerate gracefully towards MFSK
>in worsening channel conditions. If we think of OFDM as a number of
>multiplexed MFSK channels, then these channels could be made to drop out
>so that the same transmitter power becomes concentrated across a smaller
>bandwidth thus gaining SNR.

I agree: "both systems need to be developed".

>The point is, MFSK based on FFT is not bad and is very efficient compared
>to the matched correlator approach. It is also potentially upgradable in
>real-time to OFDM. The crucial issues for the MFSK DSP system are: aliasing
>and roundoff noise caused by demodulation to I/Q and decimation, FFT dynamic
>range and tuning control.

My view on the dynamic range of the FFT:

If I do a 256 point FFT, I scale my input data by 1/256 to avoid potential
overflow problem (FFT output gets real messy, when overflows come into
action !). I work with 24-bit numbers (my DSP is DSP56001).

>From the CODEC I am receiving 16 bit samples which are placed in the higher
part of the word (bits 8-23). Thus, by dividing by 256 I do not loose any bit
Infact, I can divide only by 128 if I use the $\sin(x)^2$ window
so I get 1 bit margin.

Now, I do the actual FFT. I do not know what exactly happens here
but from the simulations I could see that an error is introduced
sometimes on the lowest bit. Thus I almost keep the original
dynamic range, right ?

Well, now you may say that with passband+decimation process you _gain_
extra dynamic range ? This might be true, but I ask a question:
are you sure your passband filters do have enough out-of-band rejection
so that this extra dynamic range is "real" ?

Apropos finding the highest power:

On the FFT output I get 24-bit Is and Qs. I compute power as $I^2 + Q^2$
and I get a 48-bit number (well, maybe a 49-bit ?).
I then do 48-bit comparisons to find the peak.

By the way a question about HF coherence.
If we do not expect any coherence from HF how can we expect that a piece
of pure wave (piccolo symbol) will stay pure (or coherent in other words).
If I assume that the incoming signal's phase is random, such piece of pure
wave should dissolve and its spectrum become wider.
I suspect the phase isn't changing purely randomly and the changes
are "coherent" to a certain degree.
The question is how "coherent" they are, how much can a pure tone
dissolve on HF ?

Pawel

15. Re: [HFSIG:364] Re: Proposal for a piccolo-type modulation

From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
Date: Tue, 11 Apr 1995 15:13:37 +0100
To: hfsig@tapr.org
Cc: nash@camail.ca.nmp.nokia.com
Message-ID: <199504111414.RAA15617@noknic.nokia.com>
Rendered body type: text/plain

Pawel:

>If I do a 256 point FFT, I scale my input data by 1/256 to avoid potential
>overflow problem (FFT output gets real messy, when overflows come into
>action !). I work with 24-bit numbers (my DSP is DSP56001).

>From the CODEC I am receiving 16 bit samples which are placed in the higher
>part of the word (bits 8-23). Thus, by dividing by 256 I do not loose any bit
>Infact, I can divide only by 128 if I use the $\sin(x)^2$ window
>so I get 1 bit margin.

Well there is the advantage in using a 24 bit fixed point processor!

Pawel, the way I perform FFTs on the ADSP2101 which is only 16 bits :-(),
is to monitor the exponent of the stage results in block floating point mode,
and to re-scale the block if the bit growth is 2 bits or more, that is, I
start with a 2 bit margin. If bit growth occurs, I change the block exponent
by 2 and re-scale the entire block by this new exponent. In this way I don't
lose the precision of the input data by scaling by 1/N.

It is possible that the limiting effect of block floating point can cause
problems. If, an incorrect frequency bin has a high magnitude which is close
but just less than the correct frequency bin for the tone being received,
noise in the incorrect bin could just push the block exponent up by

one which will cause the whole block to be renormalized. Also, this noise will usually have a phase component. Since the real and imaginary data are in the same block, there is crosstalk between I/Q in terms of the block exponent. This effect worsens the probability of error at low SNRs. Typically I get a character error rate (NB 1 character consists of two symbols) of $1E-02$ at an SNR of -9dB in a AWGN channel. This is about 3dB worse than the equivalent analogue MFSK modem (eg LA1117). Some of this could be due to aliasing noise though.

Pawel:

>Well, now you may say that with passband+decimation process you _gain_
>extra dynamic range ? This might be true, but I ask a question:
>are you sure your passband filters do have enough out-of-band rejection
>so that this extra dynamic range is "real" ?

There are trade-offs between simplified decimation and heavy demodulation and decimation but with zero-IF FFT-type spectral analysis instead of quadrature matched correlators (DFT). I have found in my current MFSK modem which uses demodulation in phase quadrature to shift the passband down to 0Hz plus multiple-stage decimation, there is significant aliasing which needs sorting out. Most of this aliasing is caused by the 2nd image generated by the demodulation process aliasing. Improved input filtering would cure this.

The matched correlator approach has just one decimation stage from 7680Hz to 2560Hz. This means there is less filtering so less roundoff noise and no demodulation which means no 2nd aliasing or further roundoff noise. In tests, I have not observed any significant noise due to out of band signals aliasing. This coupled with full floating point arithmetic for post-accumulation should improve the dynamic range and indeed does for one correlator tested in isolation. I think therefore, my way of selecting the peak and synchronisation epoch is wrong rather than the detector itself. The correlator seems to offer spurious free dynamic range of an acceptable range (not sure how many dBs but it works OK in the presence of very much stronger out-of-band signals.)

Pawel:

>By the way a question about HF coherence.
>If we do not expect any coherence from HF how can we expect that a piece
>of pure wave (piccolo symbol) will stay pure (or coherent in other words).
>If I assume that the incoming signal's phase is random, such piece of pure
>wave should dissolve and its spectrum become wider.
>I suspect the phase isn't changing purely randomly and the changes
>are "coherent" to a certain degree.
>The question is how "coherent" they are, how much can a pure tone
>dissolve on HF ?

MFSK is non-coherent on HF so I use non-coherent detection. The FFT or matched quadrature correlator both work with complex numbers. (NB this is another reason why I demodulate the passband to 0Hz in the complex plane - I use a 16 point complex FFT so all 16 points convey unique information) Taking the magnitude squared just as you do is phase independent so no problem with the input signal phase shifting around.

I think it is good that we are discussing in more detail the implementation issues of these digital modems for HF. After all, for an HF channel, things like dynamic range are very important and we want to develop top-performance demodulators rather than hoping that the channel coding will make up for the deficit.

Regards
Adrian

16. Re: [HFSIG:365] Re: Proposal for a piccolo-type modulation

From: JALOCHA@chopin.ifj.edu.pl
Date: Tue, 11 Apr 1995 17:11 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <B7B3D14A80A07E48@chopin.ifj.edu.pl>
Rendered body type: text/plain

Adrian:

>Well there is the advantage in using a 24 bit fixed point processor!

I see... your DSP is only 16-bits.

Are you applying any window to your data before the FFT ?

Lack of windowing is a good reason for nearby carriers leaking into the "good" ones.

Adrian:

>Taking the magintude squared just as you do is
>phase independant so no problem with the input signal phase shifting around.

I was actually wondering, that if you take a pure tone and start shifting its phase around, the tone will no longer stay pure. If the shift rate is constant the tone stays pure but changes its frequency (so it may fall into another piccolo demodulator bin). If the shift rate (or the shift itself) is random the tone will dissolve into a set of frequencies spread around the original one. If the phase shift is purely random in time, the tone gets spread all around the band.

My (bad) intention was to trigger a discussion on the coherence on the HF. People argue that you can not use phase modulation on HF because of lack of coherence, but if you do not have any coherence then your signals get spread into an infinite band so even CW doesn't get through :-)

I guess there is still some coherence left on HF :-) and I would like to know how much of it is left. How can this missed coherence influence non-coherent systems ? For example is it worth to send piccolo with 5 Hz spacing between tones or would the HF spread such tones over more than 10 Hz so detection isn't really possible ?

Pawel

17. Re: [HFSIG:366] Re: Proposal for a piccolo-type modulation

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Tue, 11 Apr 95 13:23:06 nst
To: hfsig@tapr.org
Message-ID: <9503117976.AA797631894@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

Pawel:

I suspect that the absolute error in tuning your radio will be far greater than the offset in a tone during a symbol due to phase errors. By spec on the commercial equipment I use, the Radios are supposed to be within +-20Hz (older equipment) or +-10Hz(Newer equipment, within last 3-5 years). So if one end is +off, the other is -off, a total of +-40Hz(older) or +-20Hz(newer) may be observed. I believe the marine standards for GMDSS equipment require the modems to work with upto a 40Hz difference between ends, and the radios(new) to have less than a 10Hz error.

Also, although you probably know this, but anyway...

Careful of the definition of Coherent detection. You cannot use true PSK on HF, but you can use DPSK. i.e. you do not have a reference phase, so PSK is not possible. But if you use the previous symbol as a reference, then you can use Differential PSK (DPSK).

I suspect some confuse the limit on using coherent detection with not being able to use any form of phase modulation. Thus leading to the arguments you've heard.

In HF the Phase delay can change over time, typically considered to be only a small amount during one or two 100baud symbols. So the phase error between successive symbols is small. Of course if a symbol phase is in error (phase demodulated wrong) then the next symbol may also be demodulated wrong since it is relative to it. So often (usually?) errors occur in pairs. Maybe this should be considered in the design of error correction and interleaving code for such a modem?

Over longer periods of time phase drift/error on the HF channel becomes great (>90degrees after N symbols, N > 10 guess?). So if you locked a phase reference at the beginning of the packet and tried PSK demodulation, it would probably be well off by the end of the packet.

just my 2cents worth

Paul Russell

18. Re: [HFSIG:367] Re: Proposal for a piccolo-type modulation

From: JALOCHA@chopin.ifj.edu.pl
Date: Tue, 11 Apr 1995 18:12 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <C0546E9080A07E48@chopin.ifj.edu.pl>
Rendered body type: text/plain

Thanks Paul, very informative answer and it all clear to me:
the phase is random but it drifts slowly thus one can expect
coherence over short periods of time and this is a good reason
for differential phase modulating techniques to work.

Pawel

19. Re: Proposal for a picollo-type modulation

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Tue, 11 Apr 1995 17:19:16 +0200 (MESZ)
To: hfsig@tapr.org
Message-ID: <199504111519.RAA13918@hades>
Rendered body type: text/plain

>>With 32 carriers and 4DQPSK on each of them you would have 4000bps.

>This was exactly the number I got fascinated with.
>When it came to the actual implementation, I found out I have to space
>carriers a bit wider to reduce crosstalk between them so I came down to
>3000 bps. Then, the DQPSK did not work... only DBPSK did (but now I have
>an idea why and I will come back to QPSK) so I came down to 1500 bps
>- still not too bad.
>Timo and Joni from Finland did some tests of this modem and actual
>performance was real bad. I suspect the tuning is a serious issue here.
>Their signal levels were S3..5 at the distance of about 15 km.

Yes, tunig is the main problem. But I still think that DQPSK can be used.
If you not only use the last received symbol but also the one before that
you can win back a lot compared to coherent demodulation of QPSK. Furthermore
with 2 times differential modulation you get rid of the tunig problem.
On the other hand, you loose about 3dB in S/N for the same BER. This is
of course too much. I recently came across an interesting paper which
covers a new demodulation scheme (M.K.Simon and D.Divsalar,
"On the Implementation and Performance of Single and Double Differential
Detection Schemes", IEEE Trans. Comm. COM-40, 1992, p. 278-291
and F. Edbauer, "Bit Error Rate of Binary and Quaternary DPSK Signals
with Multiple Differential Feedback Detection", IEEE Trans. Comm. COM-40, 1992,
p. 457-460).

>>The third possibility is the application of broadband signals, OCDM
>>(= orthogonal code division multiplex). I've read through an interesting
>>article on this topic recently (J. Lindner, "Channel Coding and Modulation
>>for Transmission over Multipath Channels", to be published in
>>no. 3 AEUE, May 1995).

>Would be very nice if you could say something more on the ideas
>expressed in this article.

The basic idea behind the article is the generalization of linear block
modulation with orthogonal functions. If you use single or multicarrier
transmission - that is you use orthogonal sine waves - you have OFDM.
If you use other types of orthogonal functions - you have OCDM.

The author begins with an overview over existing modulation schemes.
The multicarrier approach is quite old and was taken to reduce
the complexity of the receiver. As no equalization is needed, simple
receiver structures are feasible. It is important to mention that
single carrier schemes have proven to be more effective, but - of course -
require very complex demodulators. I like to mention the new Rhode&Schwarz
HF modem GM857 which uses coded 8PSK on a single carrier to achieve 2k4bps in
an SSB channel. Multicarrier schemes are effective, if you use advanced
coding. The rest of the paper is about the relation of channel coding
and modulation with respect to the multipath channel.

>From previous work it is clear that you can approach the available
channel capacity with OFDM by using a method called "water pouring scheme"
provided you know exactly the channel capacity of each "subchannel". This
can be done using a back channel from the receiver to the transmitter.
I think, this can also be done using ARQ techniques.

The rest of the paper is about what you can do to achieve the maximum
channel capacity if you do not have his knowledge. For OFDM the simplest

approach is signal interleaving, you swap the mapping of symbols and sub channels. Assuming Rayleigh fading channels, the thus created virtual sub channels are also Rayleigh fading.

If you distribute the symbols over more than one sub channel of the OFDM, you get OCDM. He shows that the output of the decoder on the receiver side is now not fading at the expense of intersymbol interference between different signals.

>From a practical point of view this is quite clear, as broadband signals don't suffer that much from frequency selective fading than narrow band signals do.

The main problem is the need for maximum likelihood reception due to the intersymbol interference and thus a very complex receiver.

Alexander

```
--
*-----*
|      Alexander F. Kurpiers      |
|      Institut fuer Uebertragungstechnik      |
|      Merckstrasse 25      |
|      D-64283 Darmstadt (Germany)      |
|      Voice: +49-6151-162369      |
|      Fax  : +49-6151-165545      |
|      EMail: kurpiers@zeus.uet.e-technik.th-darmstadt.de      |
|      Hamradio: dl8aau@db0kg.#nds.deu.eu      |
*-----*
```

20. Two Types of Modems

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)
Date: Tue, 11 Apr 95 12:05:39 -0500
To: hfsig@tapr.org
Message-ID: <m0ryjNj-0002BUC@sacdm10.kelly.af.mil>
Rendered body type: text/plain

For those of us listening in on the SIG who are non-technical types, I gather that there are two types of modems being discussed here.

One is a very robust modem for worst-case conditions with a low bit rate but to ensure throughput...less than 100 BPS

The second is a high-speed modem, somewhat robust and desired to operate under average to poor conditions...about 2400 BPS.

Additionally, I am assuming that development is designed to fit on multiple platforms to include generic soundcards and external DSP devices such as the TAPR DSP device.

Someone may want to re-define/state the efforts here so new subscribers can know our direction.

On a personal note, I believe the greatest interest of this project by the amateur radio community will be the ability of the average ham to use a simple computer soundcard to send and receive data on HF at data rates of up to 2400 BPS over average to poor paths.

Keep up the fine effort.

Best Regards,

qWalt/K5YFW

21. PDM-2

From: "Johan Forrer" <FORRERJ@frl.orst.edu>
Date: Tue, 11 Apr 1995 13:47:45 PST8PDT
To: HFSIG@tapr.org
Message-ID: <735F1242AA4@frl.orst.edu>
Rendered body type: text/plain

Alexander,

Thanks for the Simon & Divsalar reference.

I learned something new today.

For those interested - DPSK derives signaling states from the difference in

phase between successive symbols. If only adjacent symbols are considered, this is also called PDM-1, however, if one take the difference spread out over 'm' several preceeding symbols, this is called PDM-m: PDM-2 when the past two symbols are used - this is also called the second deriative of the phase difference. The neat thing about this scheme is that doppler shift on the signal is cancelled in the detection process! It is stated that PDM-2, when no frequency offset is present, is some 3 - 4 dB worse than PDM-1 (i.e. regular DPSK), however, the latter suffers severely in off-frequency situations when PDM-2 still performs reasonably. Perhaps a good method for tuning or symbol synch?

PDM-m can be implemented using either the autocorrelation method (i.e. correlating the signal with itself one symbol time delayed) or by the I/Q method (shift to baseband, integrate/dump, and autocorrelation).

I could not make much sense from the Edbauer paper - perhaps Alexander could fill us in. In the same volume (COM-40), April issue, by Casas and Leung, there also is a paper on the topic of the effects of fading on OFDM - sounds somewhat familiar?

73's

Johan Forrer, KC7WW

22. computer noise

From: Glen Worstell <Glen_Worstell@notes.seagate.com>
Date: 11 Apr 95 13:15:31 EDT
To: hfsig <hfsig@tapr.org>
Message-ID: <9504112334.AA5856@worldcom-18.worldcom.com>
Rendered body type: text/plain

Hi,

I am interested in experimenting with some of the ideas presented in this forum. To that end I connected my clone 486-33 pc to my Icom 735 and PK232. The noise level was about 20 over. Even using my two meter handheld the noise is stronger than many signals.

I'd like to purchase a new low noise PC. The computer droids know less than nothing about RFI. Does anyone have any suggestions or experience with particular brands/models? Is monitor noise also an issue?

TIA,
Glen, KG0T

23. Re: [HFSIG:369] Re: Proposal for a picollo-type modulation

From: k5yfw@k5yfw.ampr.org (Walter D. DuBose - K5YFW)
Date: Tue, 11 Apr 95 18:48:08 CST
To: hfsig@tapr.org
Message-ID: <2738@k5yfw.ampr.org>
Rendered body type: text/plain

Alexander, et al.

In message <199504111519.RAA13918@hades> Alexander writes:

>

>

- > The author begins with an overview over existing modulation schemes.
- > The multicarrier approach is quite old and was taken to reduce
- > the complexity of the receiver. As no equalization is needed, simple
- > receiver structures are feasible. It is important to mention that
- > single carrier schemes have proven to be more effective, but - of course -
- > require very complex demodulators. I like to mention the new Rhode&Schwarz
- > HF modem GM857 which uses coded 8PSK on a single carrier to achieve 2k4bps in
- > an SSB channel. Multicarrier schemes are effective, if you use advanced
- > coding. The rest of the paper is about the relation of channel coding
- > and modulation with respect to the multipath channel.

Could this be something like the single tone modem referenced in MIL-STD-188-110a, change notice 2? One tone that can "do" 4800 BPS without FEC and 2400 BPS with FEC (Reed-Solomon coding).

The single tone modem signal sounds just like 3 KHz of noise... it fits into a SSB bandpass filter on most U.S. Military transceivers.

Walt

24. Re: Proposal for a picollo-type modulation

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Wed, 12 Apr 1995 08:29:04 +0200 (MESZ)
To: hfsig@tapr.org
Message-ID: <199504120629.IAA14075@hades>
Rendered body type: text/plain

>
> Could this be something like the single tone modem referenced in
> MIL-STD-188-110a, change notice 2? One tone that can "do" 4800
> BPS without FEC and 2400 BPS with FEC (Reed-Solomon coding).
>
> The single tone modem signal sounds just like 3 KHz of noise...
> it fits into a SSB bandpass filter on most U.S. Military
> transceivers.
>
> Walt
>

I just looked in the description of the modem. You are right it's
MIL-STD-188-110a.

Alexander

25. Re: [HFSIG:366] Re: Proposal for a picollo-type modulation

From: Adrian Nash <nash@camail.ca.nmp.nokia.com>
Date: Wed, 12 Apr 1995 09:18:29 +0100
To: hfsig@tapr.org
Message-ID: <199504120819.LAA02187@noknic.nokia.com>
Rendered body type: text/plain

Pawel:

>Are you applying any window to your data before the FFT ?
>Lack of windowing is a good reason for nearby carriers leaking
>into the "good" ones.

My modem deliberately uses no window. The sidelobe information is used in two ways:

- a) by taking the ratio of power in all FFT bins to that in the current peak bin, we get a useful "confidence estimate" which is used for recovering synchronisation but can also be used as a soft decision for a Viterbi Algorithm.
- b) More specifically: I have an auto-tuning algorithm which I have tested using MathCad but currently is not in the modem. This algorithm uses the adjacent frequency bins to the peak to compute with a high degree of confidence and accuracy (so far as the simulations have shown) the frequency tuning error. This can be used to track drift and Doppler shift as well as perform fine tuning after a user has coarse-tuned the signal. The same algorithm can be made to track phase changes (possibly as quick as those you mention below) by using the FFT phase information. Perhaps this algorithm could be combined with your "triangular" course tuning system to enable the modem to automatically acquire then track the signal. I feel auto-tuning is a very important issue which is essential for amateur high-performance digital modems given the range of amateur HF equipment available.

All the bins are in effect matched filters, one for each tone with no redundancy.

Pawel:

>I was actually wondering, that if you take a pure tone and start shifting
>its phase around, the tone will no longer stay pure. If the shift rate
>is constant the tone stays pure but changes its frequency (so it may fall
>into another picollo demodulator bin). If the shift rate (or the shift itself)
>is random the tone will dissolve into a set of frequencies spread around
>the original one. If the phase shift is purely random in time, the tone
>gets spread all around the band.

This reminds me of some work I did investigating the effects of fast fading on MFSK. MFSK is a bit prone to fast fades (eg fade rate >1Hz.) The effect is that the tone's energy is spread across a band with a normal (Gaussian) distribution due to the violent (eg 180deg) phase changes that occur with this type of fade. The width of the modulation depends upon the fade rate. In the case of orthogonal MFSK, the effect of the 180 degree phase change is

to create a strong component in the adjacent frequency bins which when integrated can produce an erroneous selection.

This is where perhaps I need to look at the demodulation of MFSK as a pure spectral analysis problem rather than the slightly "digital for analogue" way in which the current modem works. Finer FFT resolution with windowing as you suggest would reduce cross-talk.

Pawel:

>I guess there is still some coherence left on HF :-)) and I would like to know
>how much of it is left. How can this missed coherence influence non-coherent
>systems ? For example is it worth to send picollo with 5 Hz spacing between
>tones or would the HF spread such tones over more than 10 Hz so detection
>isn't really possible ?

Doppler shifts of 5Hz have been observed on HF circuits therefore I assume that this narrow spacing would cause problems. Also, the narrower the separation the more prone the MFSK system becomes to selective fading. If the frequency separation of two tones is made equal to the reciprocal of the differential path delay, one tone will be at a maximum when the other is at a minimum. Therefore, a typical MFSK signal with a bandwidth between 180Hz and 360Hz (single channel) affords some protection against multipath selective fading. Infact, this is apparently where MFSK scores highly over straight DPSK - it is much better in a selective fading channel. However, I think this is not the issue it was with the advent of sophisticated channel coding techniques available.

Regards
Adrian

26. Re: [HFSIG:367] Re: Proposal for a picollo-type modulation

From: JALOCHA@chopin.ifj.edu.pl
Date: Wed, 12 Apr 1995 11:34 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <51E7C25B00A092DA@chopin.ifj.edu.pl>
Rendered body type: text/plain

Paul:

>In HF the Phase delay can change over time, typically considered to be only
>a small amount during one or two 100baud symbols. So the phase error
>between successive symbols is small.

A related question: what about the group delay variation across frequency ?
I mean when I am receiving a multi-carrier signal and I find the symbol timing for one carrier can I use same timing for other carriers ?
Or should I adjust the symbol timing for every carrier separately ?

By the way, does an average SSB filter introduces a significant group delay variation ? If we use 100 bps per carrier then variation of +/- few ms is very significant.

Pawel

27. Re: [HFSIG:372] computer noise

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Wed, 12 Apr 95 09:41:43 nst
To: hfsig@tapr.org
Message-ID: <9503127977.AA797705048@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

I have found files (inc FAQs) that mention cures for RFI at:
mailing list:
send message containing word "index" in body to info@arrrl.org

try also:
ftp://ftp.cs.buffalo.edu/pub/ham-radio/
http://dingus.n5lyt.datapoint.com:80/tapr/
http://hamster.business.uwo.ca/

If anyone out there can tell me where to get QEX, QST and the other publications often referred to on this list I would appreciate it.

Paul Russell

Reply Separator

Subject: [HFSIG:372] computer noise
Author: hfsig@tapr.org at nwtsmtp
Date: 4/11/95 8:42 PM

Hi,

I am interested in experimenting with some of the ideas presented in this forum. To that end I connected my clone 486-33 pc to my Icom 735 and PK232. The noise level was about 20 over. Even using my two meter handheld the noise is stronger than many signals.

I'd like to purchase a new low noise PC. The computer droids know less than nothing about RFI. Does anyone have any suggestions or experience with particular brands/models? Is monitor noise also an issue?

TIA,
Glen, KG0T

28. Re: [HFSIG:376] Re: Proposal for a picollo-type modulation

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>
Date: Wed, 12 Apr 95 10:10:16 nst
To: hfsig@tapr.org
Message-ID: <9503127977.AA797706668@ccsmtp.udc.neweast.ca>
Rendered body type: text/plain

>Paul wrote:

>>In HF the Phase delay can change over time, typically considered to be
>>only a small amount during one or two 100baud symbols. So the phase error
>>between successive symbols is small.

>

>Pawel wrote:

>A related question: what about the group delay variation across frequency ?
>I mean when I am receiving a multi-carrier signal and I find the symbol timing
>for one carrier can I use same timing for other carriers ?
>Or should I adjust the symbol timing for every carrier separately ?

>

>By the way, does an average SSB filter introduces a significant
>group delay variation ? If we use 100 bps per carrier then
>variation of +/- few ms is very significant.

Hadn't thought about it this way, I hope its not significant. I've been allowing the windowing effect to cure/hide any variance in delay between frequencies. i.e. the symbols were pinched near their edges, so several samples of variance would have little effect at the edges. But if I'm going to be shaping over several symbols as you have suggested, I don't know anymore?!

Definitely the phase of each tone in a symbol (M-ary DPSK) is compared to the same tone in the preceding symbol, not a reference tone, as there may be different phase delays per tone (here we are comparing the phase of a single cycle of a sine wave, not the time delay of the whole symbol's tone). But if the delay differences per tone are such that the difference in propagation of each tone affects symbol timing - our modems are going to be significantly more complex - basically a bunch of independent DPSK or FSK modems running totally independently, different FFT's, DPLL's, etc per tone.

I have seen no references in anyone's designs for this yet, including what I have read about Clover and MIL-STD-188, etc. If the variance in propagation delays would affect DPSK, they would also affect FSK. It may not be optimal, but since no-one else has found it a problem, I'm going to ignore it till I find it a problem. "Ignorance is bliss" - you are trying to steal my "bliss" <g>.

But then again, alot of these designs use correlators, not FFT's. The correlation on each tone "may" track independently from the next and thus compensate for this. Whereas an FFT samples all the Tones from the same reference time-mark. Hum...?

The following are only guesses, Someone please provide factual info:

Maybe someone more knowledgable will step in, but I toss in some idle comments here. Speed through a medium of electromagnetic signals is related to the speed of light. All of our signals when modulated to RF are relatively the same frequency - whats a 1KHz tone difference when modulated onto 5MHz - 5.000 vs 5.001 - little. So I would expect most of the group phase variance to occur while the signal is in the audio spectrum, before and after modulation (or during the modulation). I'll hazard a guess that the variance in the signals propagation no more than a fraction of a cycles of the audio tone (when in RF state). If you have the group delay plots from any of your filter designs handy,

Paul Russell

Paul Russell

TIA,
Glen, KGOT

--Johan, KC7WW

31. computer noise

From: Bob Smith <rsmith@internetMCI.COM>
Date: Wed, 12 Apr 1995 12:31:21 -0500
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <01HP92FBP07695MS20@MAILSRV1.PCY.MCI.NET>
Rendered body type: text/plain

-- [From: Bob Smith * EMC.Ver #b.2.5.02] --

You may try good grounding techniques Glen. Strap everything with the braid from RG/8 then to your station ground. This cleared things up very well for me.

Date: Tuesday, 11-Apr-95 06:10 PM

From: Glen Worstell \ Internet:
(glen_worstell@notes.seagate.com)
To: hfsig@tapr.org \ Internet: (hfsig@tapr.org)
You may try good grounding techniques Tia. Strap everything with the braid from RG/8 then to your station ground. This cleared things up very well for me.

Subject: [HFSIG:372] computer noise

Hi,
I am interested in experimenting with some of the ideas presented in this forum. To that end I connected my clone 486-33 pc to my Icom 735 and PK232. The noise level was about 20 over. Even using my two meter handheld the noise is stronger than many signals.

I'd like to purchase a new low noise PC. The computer droids know less than nothing about RFI. Does anyone have any suggestions or experience with particular brands/models? Is monitor noise also an issue?

TIA,
Glen, KG0T
----- FORWARD, End of original message -----

32. Re: [HFSIG:380] ITU/TIES

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)
Date: Wed, 12 Apr 95 16:19:10 CDT
To: hfsig@tapr.org
Message-ID: <9504122119.AA04448@eagle.aud.alcatel.com>
Rendered body type: text/plain

> From hfsig@tapr.org Wed Apr 12 11:14:35 1995
> Reply-To: hfsig@tapr.org
> Originator: hfsig@tapr.org
> Sender: hfsig@tapr.org
> To: hfsig@tapr.org
> Subject: [HFSIG:380] ITU/TIES
> X-Listprocessor-Version: 6.0 -- ListProcessor by Anastasios Kotsikonas
> X-Comment: Tucson Amateur Packet Radio HF Special Interest Group
> Content-Length: 489
> X-Lines: 18
>
> Hi All,
>
> I need some help finding a CCIR document on ITU/TIES.
>
> The ITU server is on the WWW at: <http://www.itu.ch>
> though the amount of literature are overwhelming.
> Does anyone know where to look/obtain a specific issue of the
> CCIR specifications ?
>
> I know Tom McDermott has told us how to do this, though have
> not succeeded myself. I have not been able to contact Tom via
> e-mail lately. Anyone know his whereabouts?
>
> Any help/pointers would be much appreciated.
>
> 73's
>
> --Johan, KC7WW
>

OK. Here's what I did.

1. telnet gopher.itu.ch 2000
This connects you to the gopher at CCITT.
2. There's a list of choices. Use J to go down, K to go up (UNIX standard) or you can type in the number you want, and it will go there directly.
3. I selected 3 - ITU Standards, Publications, Databases, Meetings, Press...
4. I selected 2 - ITU Radiocommunications Standards
5. From the next menu, I selected 23 - Index of ITU-R Recommendations by numerical order.
6. You can then download the raw ASCII or a WORD2.0 version.

Other paths through the gopher...

3. Select 1 - ITU Telecommunications Standards.
4. Select 13 - M series Recommendations - you can look through the list of recommendation titles on line, or select the option to download a postscript version of the list.

Another path...

Select 9 - Catalog of ITU-R Recommendations
this will show how to search by keyword(s) through the recommendation,
A search on 'Propagation' turned up a number of cited references.

there are many other search engines available, and some of the text of the recommendations, etc. are available on-line by Postscript, ASCII, or WORD2.0 download.

- Tom, N5EG

2. Select 3. from the main menu - ITU Telecommunications

```
-----+-----
Tom McDermott                | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com      | represent those of Alcatel
[ ICC'96 Technical Program Secretary ] | Network Systems, Inc."
[ June 23-27, 1996, Dallas, Tx. ]    |
-----+-----
```

33. Gopher screen shots (long)

From: mcdermot@eagle.aud.alcatel.com (Tom McDermott)
Date: Wed, 12 Apr 95 16:53:53 CDT
To: hfsig@tapr.org
Message-ID: <9504122153.AA04956@eagle.aud.alcatel.com>
Rendered body type: text/plain

Here's some screen shots on the way to finding the
ITU-R M-series recommendations.....

ITU Telecom Information Exchange Services (TIES)

International Telecommunication Union (ITU)

- ```
--> 1. About ITU, TIES, ITUDOC, Copyright, What's new.../
 2. Search menu titles in ITU Gopher <?>
 3. ITU Standards, Publications, Databases, Meetings, Press.../
 4. ITU Telecommunication Standardization Sector/
 5. ITU Radiocommunication Sector/
 6. ITU Telecommunication Development Sector/
 7. ITU General Secretariat/
 8. ITU Office of the Secretary-General/
 9. TELECOM Exhibitions and Forums/
 10. Telecommunications-related topics/
 11. United Nations & international organizations/
 12. Permanent Missions based in Geneva/
 13. Other information services/
```

## ITU Telecom Information Exchange Services (TIES)

ITU Standards, Publications, Databases, Meetings, Press...

- ```
--> 1. ITU Telecommunication Standards (ITU-T Recommendations)/
    2. ITU Radiocommunication Standards (ITU-R Recommendations)/
    3. ITU Publications (July 1994 Catalogue)
    4. ITU Schedule of Meetings/
    5. ITU Press Releases/
    6. ITU Global Telecom Directory/
    7. ITU Country Codes database/
    8. Catalogue of ITU-T Recommendations/
    9. Catalogue of ITU-R Recommendations/
   10. Catalogue of ITU-R Questions/
   11. Database of ITU Software for Frequency Management/
   12. ITU/BDT Databases/
   13. ITU Constitution, Convention, .../
   14. ITU Telecommunication Terminology Database (TERMITE)/
```

ITU Telecom Information Exchange Services (TIES)

ITU Radiocommunication Standards (ITU-R Recommendations)

- ```
--> 1. Series ITU-R B0 Recommendations [Broadcasting satellite
service../
 2. Series ITU-R BR Recommendations [Sound and television
 recording../
 3. Series ITU-R BS Recommendations [Broadcasting service (sound)]/
 4. Series ITU-R BT Recommendations [Broadcasting service
 (televisi../
 5. Series ITU-R F Recommendations [Fixed service]/
 6. Series ITU-R IS Recommendations [Inter-service sharing and
 com../
 7. Series ITU-R M Recommendations [Mobile, radiodetermination,
 a../
 8. Series ITU-R PI Recommendations [Propagation in ionized media]/
 9. Series ITU-R PN Recommendations [Propagation in non-ionized
 med../
 10. Series ITU-R RA Recommendations [Radioastronomy]/
 11. Series ITU-R S Recommendations [Fixed satellite service]/
 12. Series ITU-R SA Recommendations [Space applications]/
 13. Series ITU-R SF Recommendations [Frequency sharing between the
 ../
 14. Series ITU-R SM Recommendations [Spectrum management techniques]/
 15. Series ITU-R SNG Recommendations [Satellite news gathering]/
 16. Series ITU-R TF Recommendations [Time signals and frequency
 sta../
 17. Series ITU-R V Recommendations [Vocabulary and related
 subject../
 18. List of ITU-R Recommendations available electronically to TIES
```

reg../  
19. Table of ITU-R Recommendations (1994), Resolutions & Questions  
ava../  
20. HOW-TO-ORDER ITU-R Recommendations/  
21. Distribution of Recommendations in force as from 2 September 1994/  
22. Index of ITU-R Recommendations SERIES/  
23. Index of ITU-R Recommendations by numerical order/  
24. 1994 Volumes of ITU-R Recommendations/

Press ? Help, q Quit, u up a menu, D Download

Page: 1/1

ITU Telecom Information Exchange Services (TIES)

Index of ITU-R Recommendations SERIES

--> 1. [ASCII Lang: EFS Bytes: 3492 Updated: 94-01-06 ITU-6701]  
2. [WINWORD2.0 Lang: EFS Bytes: 7869 Updated: 94-09-06 ITU-669..  
<Bin>

Press ? Help, q Quit, u up a menu, D Download

Page: 1/1

ITU Telecom Information Exchange Services (TIES)

List of ITU-R M-Series Recommendations in force

--> 1. [ASCII Lang: EFS Bytes: 62628 Updated: 94-11-16 ITU-6712]  
2. [WINWORD2.0 Lang: EFS Bytes: 54375 Updated: 94-11-15 ITU-66..  
<Bin>

---

|                                        |                            |
|----------------------------------------|----------------------------|
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |

---

### 34. Re: [HFSIG:380] ITU/TIES

From: erik@ve7mdl.ampr.org  
Date: Wed, 12 Apr 95 19:07:06  
To: hfsig@tapr.org  
Message-ID: <12157@ve7mdl.ampr.org>  
Rendered body type: text/plain

CCIR is now called ITU-R. You can get a list of the documents via email.  
I think the email address is 'itudoc@itu.ch'. Try putting 'info' or  
'index' in the first line of the message.

If it does not work, try sending me a message at 'erik@mda.ca' and I will  
dig out the detailed instructions at work.

73 de VE7MDL

....Erik.

### 35. Optimal OFDM pulses

From: mcdermot@eagle.aud.alcatel.com (Tom McDermott)  
Date: Thu, 13 Apr 95 09:02:40 CDT  
To: hfsig@tapr.org  
Message-ID: <9504131402.AA05630@eagle.aud.alcatel.com>  
Rendered body type: text/plain

I came across an interesting reference:

"Optimal Finite Duration Pulses for OFDM", A. Vahlin & N. Holte,  
Globecom '94, pg. 258-262, paper #7.7

It discusses the generation of a pulse that is finite in duration, yet has  
good spectral properties for OFDM systems. Three examples listed look  
pretty good. Unfortunately, there is no closed-form solution to the  
minimization effort. It requires solving 20 simultaneous equations looking  
for optimal coefficients of a spheroidal wave function I've never heard of  
(and  
is not documented in the paper). The author provides the coefficients for  
2T, 3T, and 4T length pulses (where h(t) is zero outside the interval), and  
the very good spectrum for the pulses. However, without the PSWF,  
I cannot numerically regenerate h(t). The article has some references to  
the BSTJ, back about 1960 when this wave function was defined.

There are some other references in the article pointing to more recent IEEE  
Transactions on Communications addressing the subject, but I don't have the  
article with me. If there is any interest, I'll post the references.

- Tom, N5EG

---

|                                        |                            |
|----------------------------------------|----------------------------|
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |

---

---

### 36. Re: [HFSIG:378] Re: Proposal for a piccolo-type modulation

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)  
Date: Thu, 13 Apr 95 09:23:44 CDT  
To: hfsig@tapr.org  
Message-ID: <9504131423.AA05641@eagle.aud.alcatel.com>  
Rendered body type: text/plain

Paul Russell writes:

> Definitely the phase of each tone in a symbol (M-ary DPSK) is compared to  
the  
> same tone in the preceding symbol, not a reference tone, as there may be  
> different phase delays per tone (here we are comparing the phase of a single  
> cycle of a sine wave, not the time delay of the whole symbol's tone). But if  
the  
> delay differences per tone are such that the difference in propagation of  
each  
> tone affects symbol timing - our modems are going to be significantly more  
> complex - basically a bunch of independent DPSK or FSK modems running  
totally  
> independently, different FFT's, DPLL's, etc per tone.

Most of the literature on HF channel simulation has shown/assumed  
that the channel is non-dispersive out to 4 khz bandwidth. This means that  
the propagation velocity is not frequency-dependent. This assumption  
breaks down at greater bandwidth. Thus, symbol-timing can generally be  
derived for a group of carriers. However, the symbol duration must be long  
enough that multi-path distortion does not move an individual pulse or  
channel very much with respect to the recovered symbol timing. This  
timing is usually in the range 50-200 baud on many HF circuits.

- Tom, N5EG

|                                        |                            |
|----------------------------------------|----------------------------|
| -----+-----                            |                            |
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |
| -----+-----                            |                            |

---

### 37. Re: [HFSIG:382] Re: ITU/TIES

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Thu, 13 Apr 1995 10:42:53 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <762DDAD64A6@frl.orst.edu>  
Rendered body type: text/plain

Greetings,

Thanks for all the help with ITU/TIES.

73's

Johan, KC7WW

---

### 38. DSP Developer Needed

From: "Thomas E. O'Boyle" <tomob@netcom.com>  
Date: Fri, 14 Apr 1995 12:59:55 -0700 (PDT)  
To: HFSIG <hfsig@tapr.org>  
Message-ID: <Pine.3.89.9504141237.A2107-0100000@netcom15>  
Rendered body type: text/plain

Greetings to members of the HFSIG!

We are a small company (six people) located in Chicago that is working  
to finish development of a commercial HF data product. We have a  
targeted niche application in mind, and have received one patent and another  
is pending. Our design is based on a TMS320 series DSP. We have modeled  
the specific modulation approach that we intend to use against a simulation  
of multi-path and Gaussian noise and believe that we are on the right track.  
We are looking for an additional full-time engineer for this project.  
Qualifications are microcontroller and DSP experience and a love of radio.  
This might be the chance to enhance your career while working on a project  
related to your hobby. Competitive pay and benefits; informal, hardworking,  
entrepreneurial environment. For more info, fax a letter and resume to

(312) 463-9738, or respond by private E-mail.

Thanks to Johan Forrer for allowing us to post this message.

73 de Tom, N9GUN.

tomob@netcom.com

---

### 39. DSP Developer Needed

From: "Thomas E. O'Boyle" <tomob@netcom.com>  
Date: Fri, 14 Apr 1995 12:55:11 -0700 (PDT)  
To: HFSIG <hfsig@tapr.org>  
Message-ID: <Pine.3.89.9504141215.A2107-0100000@netcom15>  
Rendered body type: text/plain

\*\*

tomob@netcom.com

---

### 40. my web page

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 14 Apr 1995 13:56:45 -0700  
To: hfsig@tapr.org, tcp-group@ucsd.edu  
Message-ID: <199504142056.NAA21549@servo.qualcomm.com>  
Rendered body type: text/plain

I've finally gotten around to creating a web page. There's not a lot of technical content yet, but I intend to update it with pointers to my various papers and software projects.

The URL is

<http://www.qualcomm.com/people/pkarn>

--Phil

---

### 41. PACK&\

From: jra@ag9v.ampr.org  
Date: Sat, 15 Apr 1995 14:00:09 -0400 (EDT)  
To: aprssig@tapr.org, hfsig@tapr.org, dsp93sig@tapr.org  
Message-ID: <199504151800.OAA05660@ag9v.ampr.org>  
Rendered body type: text/plain

Attending the 1995 Dayton Hamvention? Time's running out to register for PACK\*BASH!

What?

- A new event for the digitally-inclined ham, featuring:
- \* Buffet dinner (with cash bar)
  - \* Dewayne Hendricks, WA8DZP, speaking on PCS, SS, FCC, and other acronyms
  - \* Bob Bruninga, WB4APR, speaking on the Future of APRS
  - \* Prize raffle
  - \* TAPR special interest group meetings
  - \* "Birds of a Feather" gatherings

When?

Friday evening, April 28, 1995  
Dinner starts at 7:00 pm  
Speaker at 7:45  
Meetings to begin after the speaker

Where?

TJ's Restaurant  
2462 Dryden Road, Dayton  
(Dryden Road exit, I75 -- across from the Holiday Inn)

How?

Dinner requires advance registration and payment BY APRIL 24th, 1995. Seating for dinner only is limited, so register early! The cost is \$16.00 per person, tax and tip included.

All amateurs are welcomed to attend, enjoy the speaker, and participate in the meetings, although only those purchasing a dinner can eat.

To register, contact:  
PACK\*BASH



c/o TAPR  
8987-309 E. Tanque Verde Road #337  
Tucson, AZ 85749-9399

Phone: (817) 383-0000  
Fax: (817) 566-2544  
Internet: TAPR@TAPR.ORG

Visa/Mastercard Accepted

Who?

PACK\*BASH is co-sponsored by TAPR -- Tucson Amateur Packet Radio,  
the national leader in digital communication -- and the Miami  
Valley FM Association, Dayton's packet radio club.

For further information, contact TAPR, or email packbash@ag9v.ampr.org or  
BASH@N8ACV.#DAY.OH.USA.NOAM

--

John Ackermann AG9V  
Internet: jra@ag9v.ampr.org  
Packet: AG9V@N8ACV.#DAY.OH.USA

---

## 42. HFSIG activities

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 18 Apr 1995 11:21:35 PST8PDT  
To: HFSIG@tapr.org  
Message-ID: <56CCA54471@frl.orst.edu>  
Rendered body type: text/plain

Hi All,

A brief update: There are considerable amount of activity going on behind  
the scenes - this mostly has to do with DSP specific stuff. Adrian, G4ZHZ,  
and Pawel, SP4VRC, both are making good progress.

One reality that we have to deal with is that different DSP platforms are  
being used - understandable for good reasons. I am trying to stay involved  
by cross-porting where possible. For this reason, I have been working at  
porting some of the Finnish group's (AleF Null) code. This code runs on  
a Motorola 56K-based card and I am porting this to the new low cost  
Motorola EVM56K, which some of you may or not know about. I have had some  
good success and this effort now gives access to some great software  
contributions from the AleF Null group, in particular, Jarkko Vuori,  
OH2LNS. Pawel also has a wealth of good ideas that runs on M56K - I am just  
starting to appreciate some of it.

A bit more about the EVM56K: It is a small card, about the size of a TI  
DSK, has a 56002 at 40 MHz, and 3 x 32K x 8 zero wait static ram. It uses a  
Crystal Semi CS4215 CODEC and in addition, has an optional FLASH EPROM that  
allows for making it a free-standing unit. The debugger that comes  
with it uses the 56002's built in hardware debugger, called the "OnCE" port,  
and is a powerful and non-intrusive hardware/software tool. It is different  
from the TI DSK in one respect, that the OnCE does not exist as part of the  
DSP's application program. The debugger, however, operates through the PC's  
serial line at 19.2K - a bit slow for serious development work and too  
slow for doing some of the shared DSP/host applications that we have in  
mind. If you are a Motorola fan, this is a neat piece of hardware.

For the moment, and at least for the foreseeable future, there will be  
emphasis on the PSA-type sound cards and their closely-coupled architecture  
with the PC's resources, as well as the M56K architecture. Stay tuned to  
these projects if you are interested in low-level developments.

I hope to be doing some work on the DSP-93 one of these days - though I  
am in great need of a faster intercommunication channel between the host  
and the DSP-93. If anyone has some ideas, please contact me in this regard.

Keep up the good work, 73's

--Johan, KC7WW

---

## 43. Re: [HFSIG:392] HFSIG activities

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Tue, 18 Apr 95 15:19:01 -0500  
To: hfsig@tapr.org  
Message-ID: <m0s1Jjh-0002BvC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

Johan,

For those of us running high speed TCP/IP on the AmprNet, I don't think anyone has found anything faster than the PI card. Its gonna be hard to beat a DMA card...a card plugged into the motherboard's bus, for speed. I believe that is part of the reason you originally went with the Soundwave and Cardinal cards. The Soundwave 32 can be bought on the surplus tables for less than \$75 now.

I see great benefit in off-loading some functions but keep one thing in mind...if we are ever to gain wide acceptance of this effort, we need to keep the effort within the confines of what computing power that is generally available to hams without too much additional hardware. This is why the Baycom is so popular.

Keep up the good work.

73 to all.

Walt

In your message of 18 Apr 1995 at 1335 CDT, you write:

> Hi All,

>

> A brief update: There are considerable amount of activity going on behind the scenes - this mostly has to do with DSP specific stuff. Adrian, G4ZHZ, and Pawel, SP4VRC, both are making good progress.

>

> One reality that we have to deal with is that different DSP platforms are being used - understandable for good reasons. I am trying to stay involved by cross-porting where possible. For this reason, I have been working at porting some of the Finnish group's (Alef Null) code. This code runs on a Motorola 56K-based card and I am porting this to the new low cost Motorola EVM56K, which some of you may or not know about. I have had some good success and this effort now gives access to some great software contributions from the Alef Null group, in particular, Jarkko Vuori, OH2LNS. Pawel also has a wealth of good ideas that runs on M56K - I am just starting to appreciate some of it.

>

> A bit more about the EVM56K: It is a small card, about the size of a TI DSK, has a 56002 at 40 MHz, and 3 x 32K x 8 zero wait static ram. It uses a Crystal Semi CS4215 CODEC and in addition, has an optional FLASH EPROM that allows for making it a free-standing unit. The debugger that comes with it uses the 56002's built in hardware debugger, called the "OnCE" port, and is a powerful and non-intrusive hardware/software tool. It is different from the TI DSK in one respect, that the OnCE does not exist as part of the DSP's application program. The debugger, however, operates through the PC's serial line at 19.2K - a bit slow for serious development work and too slow for doing some of the shared DSP/host applications that we have in mind. If you are a Motorola fan, this is a neat piece of hardware.

>

> For the moment, and at least for the foreseeable future, there will be emphasis on the PSA-type sound cards and their closely-coupled architecture with the PC's resources, as well as the M56K architecture. Stay tuned to these projects if you are interested in low-level developments.

>

> I hope to be doing some work on the DSP-93 one of these days - though I am in great need of a faster intercommunication channel between the host and the DSP-93. If anyone has some ideas, please contact me in this regard.

>

>

> Keep up the good work, 73's

>

> --Johan, KC7WW

>

---

## 44. DSP Platforms

From: "Johan Forrer" <FORRERJ@frl.orst.edu>

Date: Wed, 19 Apr 1995 10:37:35 PST8PDT

To: HFSIG@tapr.org

Message-ID: <6E1137374D@frl.orst.edu>

Rendered body type: text/plain

Hi Walt,

Just some further clarification on DSP platforms.

For the kind of experimental work that we are doing, we need:

- a) sufficient DSP cycles (the more the better),
- b) sufficient bandwidth between the DSP platform and the PC.

The latter is not 100% required, but allows for exploration of closely coupled DSP and PC algorithms. A good example would be to let the DSP do real-time convolution/correlation/FFT's and use Phil's Viterbi algorithm, running on the PC, to steer the DSP algorithms to achieve synchronization and error-control. Eventually, however, one would like that to be part of a free-standing system, but sure helps a lot if you have access to working code on a PC while you are in development phases.

DMA is no real issue, however, a serial device working in the order of 115k or better, alternatively a high-speed parallel path is workable. DSP sound cards happen to offer this kind of interface, and with a bit of additional work, the DSP-93 or EVM56K can be made to do this too.

It is my opinion that the DSP platform should not distract innovation, actually it adds a little "spice" to the effort - it would have been ideal if we all used the same hardware, but keep in mind that some of our ideas and contributions come from folks that do not have the financial means or access to technology that we take for granted. I think the appropriate thing to do is to let each use whatever they feel comfortable with - if it works, that is all that matters for the moment. However, in order to maintain some level of continuity, someone has to take on the task to port/translate ideas for further evaluation by others. The time will come when the technical details will be written up. I foresee a number of opportunities that will arise from that and help with that effort will be needed then. I guess it will be a while before we will have "turn key" systems - in the mean time, those that wish to be involved will have to roll up their sleeves and dig into the implementations themselves - it's no simple task, but very educational, that is if one wants to learn about how this technology works. These are exciting times.

I forgot to mention the cost of the new EVM56K - it's only \$150, really good value for what you get - available from authorized Motorola dealers. I will shortly make available a software package that makes the EVM56K compatible with the Alef Null project and thus allows access to their existing software (LPC, various noise filters, 1200 Bell 202, 9600 G3RUH, and DSP code for KISS/HDLC interfaces).

We keep on looking for help with this SIG - someone to help write quarterly reports for PSR and help with running the SIG. Thanks.

Remember also that TAPR membership supports this forum - if you are not yet a member of TAPR, kindly consider joining.

73's - keep up the good work.

Johan, KC7WW  
April, 1995

---

## 45. RE: [HFSIG:394] DSP Platforms

From: Murphree Michael A <mamurphree@space.honeywell.com>  
Date: Wed, 19 Apr 95 14:57:00 PDT  
To: hfsig <hfsig@tapr.org>  
Message-ID: <2F958746@smtpdos.space.honeywell.com>  
Rendered body type: text/plain

>I forgot to mention the cost of the new EVM56K - it's only \$150,  
>really good value for what you get - available from authorized Motorola  
>dealers. I will shortly make available a software package that makes the  
>EVM56K compatible with the Alef Null project and thus allows access to  
>their existing software (LPC, various noise filters, 1200 Bell 202, 9600  
>G3RUH, and DSP code for KISS/HDLC interfaces).

Texas Instruments is again advertising their 320X DSK board for \$99 as well. They do not identify exactly which processor the 2X is...

Mike N4CNW

---

## 46. TAPR @ Dayton '95

From: Greg Jones <gjones@tenet.edu>  
Date: Sun, 23 Apr 1995 21:39:38 -0500 (CDT)  
To: bbssig@tapr.org (BBS SIG mailing), netsig@tapr.org (NETSIG mailing),  
hfsig@tapr.org (HF SIG mailing), aprssig@tapr.org  
Message-ID: <199504240239.VAA14970@Alice-Thurman.tenet.edu>  
Rendered body type: text/plain

## TAPR Activities at Dayton '95

Dayton is almost upon us and here is the list of TAPR activities and events happening during Dayton. We hope to see lots of folks around and please stop by the TAPR booth and say hi!

### TAPR Booth

The booth is located at the same location as the last several years. We should have most items and kits available and another big membership offer will be available to new members! Come by and say hi to the TAPR gang and if you want, help out behind the table. The more the merrier.

Regional Groups Note: If you are with a regional digital group, please stop by and chat with Greg Jones, WD5IVD about how TAPR can do things with your regional group. There will also be a place at the booth to place regional materials, so if you have something to hand out, drop it off.

### Friday - TAPR Digital Forum

|                 |                                                              |
|-----------------|--------------------------------------------------------------|
| 1:00pm - 1:45pm | Introduction to Digital Communications<br>Greg Jones, WD5IVD |
| 1:45pm - 2:00pm | TAPR Update<br>John Ackerman, AG9V                           |
| 2:00pm - 2:30pm | Layer 1 and Radio Issues<br>Mel Whitten, K0PFX               |
| 2:30pm - 3:00pm | TAPR/AMSAT DSP-93 Project Update<br>Bill Reed, WD0ETZ        |
| 3:00pm - 3:30pm | TCP/IP Issues and Trends<br>Barry McLaren, VE3JF             |
| 3:30pm - 3:45pm | BBS Issues and Trends<br>Barry Buelow, WA0RJT                |
| 3:45pm - 4:00pm | HF Digital Communications Trends                             |
| 4:00pm - 4:30pm | APRS<br>Bob Bruninga, WB4APR                                 |
| 4:30pm - 5:30pm | TAPR APRS SIG<br>Keith Sproul, WU2Z                          |

### PACK\*BASH T95 - Friday Evening

A new event for the digitally-inclined ham, featuring:

- \* Buffet dinner (with cash bar)
- \* Nationally-known speakers holding forth on current topics:
  - \* Bob Bruninga, WB4APR (Author of APRS)
  - \* Dewayne Hendricks, WA8DZP
- \* Prize raffle
- \* Lots of Informal Discussion
- \* TAPR NET SIG and a second APRS SIG meeting

#### When?

Friday evening, April 28, 1995  
Dinner starts at 7:00 PM, speaker at 7:45 PM  
Everyone is invited to attend, but dinner is only available to those with reservations.

#### Where?

TJ's Restaurant, 2462 Dryden Road, Dayton  
(Dryden Road exit, I75 -- across from the Holiday Inn)  
(Maps available at the TAPR booth)

#### How?

Dinner requires advance registration (by April 24th)  
Come by the TAPR booth at Dayton to check on additional space.  
Everyone is invited to come, but dinner is only available to those with reservations.  
Dinner cost is \$16.00 per person, tax and tip included.

#### To register, contact:

TAPR (tapr@tapr.org)  
Phone: (817) 383-0000, Fax: (817) 566-2544

#### Who?

PACK\*BASH is co-sponsored by TAPR -- Tucson Amateur Packet Radio,

the national leader in digital communication -- and the Miami Valley FM Association, Dayton's packet radio club.

(This is a replacement of the McNasty's event)

Saturday - BBS SIG Meeting

-----  
The TAPR BBS SIG will meet Saturday night at 7:00pm

Where: Radison Hotel - Premier Room  
The Radison is located near the intersection of I-75 and Needmore Rd. This is not far from Hara Arena.

New feature this year - CHAIRS !!!

This is a great opportunity to meet with fellow BBS Sysops and discuss current events and plans for the future. TAPR membership is not a requirement, this event is open to the public.

For more information, stop by the TAPR booth at HARA.

-----  
Tucson Amateur Packet Radio (tapr@tapr.org)  
8987-309 E Tanque Verde Rd #337 \* Tucson, Az \* 85749-9399 \* 817-383-0000

---

#### 47. Re: [HFSIG:395] RE: DSP Platforms

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>  
Date: Mon, 24 Apr 95 09:25:38 nst  
To: hfsig@tapr.org  
Message-ID: <9503247987.AA798740923@ccsmtp.udc.neweast.ca>  
Rendered body type: text/plain

>>>I forgot to mention the cost of the new EVM56K - it's only \$150,  
>really good value for what you get - available from authorized Motorola  
>dealers. I will shortly make available a software package that makes the  
>EVM56K compatible with the Alef Null project and thus allows access to  
>their exsisting software (LPC, various noise filters, 1200 Bell 202, 9600  
>G3RUH, and DSP code for KISS/HDLC interfaces).

Texas Instruments is again advertising their 3202X DSK board for \$99  
as well. They do not identify exactly which processor the 2X is...

Mike N4CNW<<

I'm fairly certain you can get the TMS320C5X DSK for \$99. Its a fixed point DSP  
upgrade from the TMS320C26 (C25+bootloader) of the other DSK.

Paul Russell

---

#### 48. RE: [HFSIG:397] RE: DSP Platforms

From: Murphree Michael A <mamurphree@space.honeywell.com>  
Date: Mon, 24 Apr 95 08:56:00 PDT  
To: hfsig <hfsig@tapr.org>  
Message-ID: <2F9BC9E4@smtpdos.space.honeywell.com>  
Rendered body type: text/plain

>>>>I forgot to mention the cost of the new EVM56K - it's only \$150,  
>>really good value for what you get - available from authorized Motorola  
>>dealers. I will shortly make available a software package that makes the  
>>EVM56K compatible with the Alef Null project and thus allows access to  
>>their exsisting software (LPC, various noise filters, 1200 Bell 202, 9600  
>>G3RUH, and DSP code for KISS/HDLC interfaces).

>  
>Texas Instruments is again advertising their 3202X DSK board for \$99  
>as well. They do not identify exactly which processor the 2X is...

>  
>Mike N4CNW<<

>  
>  
>  
>  
>I'm fairly certain you can get the TMS320C5X DSK for \$99. Its a fixed point  
>DSP  
>upgrade from the TMS320C26 (C25+bootloader) of the other DSK.  
>  
>Paul Russell

>

Actually I have two of the 320C50 DSKs that I bought when they came out. The current ad is indicating the 20 series family though... I never saw any work done to translate some of the code available for the '20 DSK to the '50 DSK. I started to do this myself some time ago, but have not found sufficient time to finish it.

Mike N4CNW

---

# May 1995

Original source file: hfsig.9505

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-05/TAPR-HF\\_1995-05.txt](https://archive.org/download/TAPR-HF_1995-05/TAPR-HF_1995-05.txt)

## 1. coding demo on WWW

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 19 May 1995 19:09:07 -0700  
To: tcp-group@ucsd.edu, hfsig@tapr.org  
Cc: hams@qualcomm.com  
Message-ID: <199505200209.TAA24644@servo.qualcomm.com>  
Rendered body type: text/plain

I have added a page to my WWW collection that I believe vividly demonstrates the potential of forward error correction (FEC) coding in power-limited amateur radio communications.

The page contains links to several synthesized audio files I've created that contain simulated noise-corrupted CW and BPSK signals at various signal-to-noise ratios. The BPSK signals further represent S/N ratios with and without FEC. Some of the coded BPSK signals are completely buried in the noise to the human ear, yet they could still be recovered with a good PSK modem and sequential decoder.

The page is <http://www.qualcomm.com/people/pkarn/fecdemo/index.html>.

You'll need a computer capable of playing 64kb/s mu-law PCM audio files.

--Phil

---

## June 1995

Original source file: hfsig.9506

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-06/TAPR-HF\\_1995-06.txt](https://archive.org/download/TAPR-HF_1995-06/TAPR-HF_1995-06.txt)

## 1. some questions

From: "Kontola Ilkka (NMP)" <kontola@nmpies01tr.nmp.nokia.com>

Date: Thu, 15 Jun 95 13:41:00 gmt

To: hfsig <hfsig@tapr.org>

Message-ID: <2FE01D36@ms-smtp-gw.nmp.nokia.com>

Rendered body type: text/plain

This is my first posting to the list so please forgive me my ignorance.

What kind of delay spread there is on NVIS paths? The ground wave could be one path but is there ionospheric multipath modes in NVIS (of course in addition to the \_wanted\_ ionospheric refraction/reflection)?

I have no hands-on experience on HF digital modulations - exept CW ;-).

In literature there are often casual comments "... on 3.5 MHz packet the multipath is very severe problem ...." and so on. Are the article/book authors referring to DX or multi-hop paths or shorter ranges?

Pointers to existing DSP5600X classic modem codes (RTTY, AMTOR, PACTOR, 300 bps FSK / "HF packet") as well as a pointer to a PC PACTOR program (requiring only a modulator and demodulator) would be appreciated!

I would like to get some feeling about the classic HF digital modes but I dont want to spend much time coding soon-to-become-obsolete modems and protocol software.

Is anyone working with adaptive equalizer -based HF modem?

BTW I have two DSP56000-family development systems:

- Ariel PC-56 (an ISA card with a 14-bit codec and a 20 MHz 56001)
- DSP56002EVM (two days old.... not much experience yet)

73.

Ilkka OH3NJC

\*\*\*\*\*

Ilkka Kontola

E-Mail: ilkka.kontola@nmp.nokia.com

## Nokia Mobile Phones

## Cellular Data

Tampere, Finland

## 2. Re: [HFSIG:400] some questions

From: "Johan Forrer" <FORRERJ@frl.orst.edu>

Date: Thu, 15 Jun 1995 08:55:52 PST8PDT

To: [hfsig@tapr.org](mailto:hfsig@tapr.org)

Message-ID: <27F55CE0C08@frl.orst.edu>

Rendered body type: text/plain

Dear Ilkka,

To be honest, I don't really know what "NVIS" means - can anyone help?

Just for interest sake, you may find it useful to browse the files in  
ftp.tapr.org /tapr/SIG/hfsig/upload

There are quite a bit on HF propagation; as related to it's affects on digital transmissions.

You will also find two Pactor programs there, one that I wrote "PSATOR/PSA-PACTOR" for the sound card DSP, and another one by Tom Sailer, HB9JNX, that will work with a number of different types of modems, including something that you can put together on your new EVM56002.

I have ported the Alef Null kernel to the EVM56002 and thus allows you to run any of the codes that run on the DSPCARD4. This includes 1200 baud BELL 202 and 9600 baud PAM (G3RUH) style modems. The DSP kernel includes all HDLC and KISS interfaces, so all you need to have is the EVM and one of the flavors of NOS/KA9Q on your PC. It's really neat stuff. For the HF side of things, I have some HF modems for the EVM as well.

Using the ported kernel, you will also be able to participate in the



experimental high speed HF modem work. There also are a number of excellent noise reduction programs using Wiener, LMS and correlation-based algorithms. The latter is by Pawel (SP9VRC) and is a joy to use on CW.

I am working on putting together the schematic for my version of the EVM's radio interface and will post the complete package (with the kind permission of the Alef Null group) on the net. Watch for the announcement if you are interested.

Think this answers most of your questions. Feel free to e-mail me directly about the EVM56002.

73's

--Johan, KC7WW  
(email: forrerJ@ucs.orst.edu)

---

### 3. New Clover board

From: Jack Taylor <n7oo@hereford.ampr.org>  
Date: Thu, 15 Jun 1995 10:41:18 -0700  
To: hfsig@tapr.org  
Message-ID: <199506151745.MAA09383@dingus.n5lyt.datarace.com>  
Rendered body type: text/plain

Page 177 of the June 95 QST has an announcement from HAL on their new \$395 DSP modem, the P38. Understand this uses one of the TI chips and comes with software for clover, pactor, amtor, baudot and ascii. If packet could be ported to it, one would have a near ideal modem!

73 de Jack

---

### 4. Re: [HFSIG:402] New Clover board

From: Greg Jones <gjones@tenet.edu>  
Date: Thu, 15 Jun 1995 13:06:47 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199506151806.NAA06308@beall.tenet.edu>  
Rendered body type: text/plain

Very similar to the DSP-93 except that it has a 68000 processor on it and is a plug in card.

So - to port code you have to program the 68000 and TI chip. Makes it much more fun.

Bill Henry intro the unit at Dayton. Interesting unit.  
You can think of it as a scaled down clover board.

I think Bill or someone from HAL is on the mail list - they could probably go into it in a lot more detail.

Cheers - Greg

According to Jack Taylor:

>  
> Page 177 of the June 95 QST has an announcement from HAL on their new \$395  
> DSP modem, the P38. Understand this uses one of the TI chips and comes with  
> software for clover, pactor, amtor, baudot and ascii. If packet could be  
> ported to it, one would have a near ideal modem!  
>  
> 73 de Jack  
>  
>

---

### 5. Re: [HFSIG:402] New Clover board

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)  
Date: Thu, 15 Jun 95 13:56:38 CDT  
To: hfsig@tapr.org  
Message-ID: <9506151856.AA02000@eagle.aud.alcatel.com>  
Rendered body type: text/plain

> From hfsig@tapr.org Thu Jun 15 13:16:07 1995  
> Reply-To: hfsig@tapr.org  
> Page 177 of the June 95 QST has an announcement from HAL on their new \$395  
> DSP modem, the P38. Understand this uses one of the TI chips and comes with  
> software for clover, pactor, amtor, baudot and ascii. If packet could be  
> ported to it, one would have a near ideal modem!  
>

> 73 de Jack  
>  
>

I talked with Bill Henry for a few minutes at Hamcom about the P38 board,  
he said that the 320C25 runs at 50 Mhz., and it uses the 68360 Motorola processor.  
The 68360 has a 68020 core, and has several HDCL and other serial channels built in, plus DMA controllers -- the 68360 is a very powerful chip (we use them at work).  
The HAL P38 board will not work at the two highest speeds of CLOVER II. The T.I. chip does not have the capability of the Motorola 56000 that is used on the original CLOVER board. Many of the tricks in the 56000 are what made Clover run fast..

- Tom, N5EG

|                                        |                            |
|----------------------------------------|----------------------------|
| -----+-----                            |                            |
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |
| -----+-----                            |                            |

---

## 6. Re: [HFSIG:403] Re: New Clover board

From: elvey@hal.com (Dwight Elvey)  
Date: Thu, 15 Jun 1995 13:29:53 -0700 (PDT)  
To: hfsig@tapr.org  
Message-ID: <9506152029.AA07941@civic.hal.com>  
Rendered body type: text/plain

Greg Jones <gjones@tenet.edu> wrote:

>  
> I think Bill or someone from HAL is on the mail list - they could probably  
> go into it in a lot more detail.  
>  
> Cheers - Greg

Hi All

I hope I'm not the one that Greg is refering to. I work for Hal Computer Systems. Other then our software company in Austin we are making a high end work station. No modems 8-)

I'm not sure if I posted this stuff before but I have done some stuff that might be of interest. I have used a DigiCom Systems modem, called the Connection 14.4+ SoftModem, as my development platform. The board has a MAFE that is primarily used for phone modems. Normally the MAFE has a phase lock loop that syncs to the various baud rates. It also has the ability to do various asynchronous sample rates up to 38.8K. This is the way I use it. The board has either a 2115 or a 2101 on it depending on the model and it has a full complement of loadable RAM in the program and data memory ( 14Kx24 & 14Kx16 ). These boards are still sold by a few people at about \$60 to \$80 each or one could get them used for about \$35. I have written some tools for these, including a binary loader, that are available from ftp taygeta.oc.nps.navy.mil under /pub/Forth/Reviewed/DSP/ as dspkit.zip and hfrfax.zip. The hfrfax was a radio fax demodulator that I made using this platform. The assembler that I used for this was one of my own invention and may not be to all's liking.

I also have one of the ADI's EZKIT Lites that I have written a DOS loader and uploader for. These are available from ftp ftp.analog.com under /pub/ as ezup1.zip and ezld1.zip. I intend to make a translator to change the ADI's \*.EXE format into a format that can be loaded into the Connection board. This will allow one to work in the assembler that came with the EZKIT to write code to work on the Connection board. Only thing that one would have to watchout for would be not to use the few instructions that are only on the 2181 that comes with the kit.  
Dwight

---

## 7. Re: [HFSIG:401] Re: some questions

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Thu, 15 Jun 1995 20:42:45 +0100  
To: hfsig@tapr.org

Message-ID: <199506152129.AA19041@felix.dircon.co.uk>

Rendered body type: text/plain

>Dear Ilkka,

>

>To be honest, I don't really know what "NVIS" means - can anyone help?

>

It stands for Near Vertical Incidence Skywave and means the signal goes almost vertically up. The idea is to cut out the skip zone for tactical H.F communications rather than for strategic systems. It's all the thing in military circles at the moment.

Regards Charles G4GU0.

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 8. [HFSIG:401] Re: some questions

From: ALAN\_BLOOM@HP5300.desk.hp.com  
Date: Thu, 15 Jun 95 15:40:00 -0700  
To: hfsig@tapr.org  
Message-ID: <"d0f,IAR0000000000\*"@MHS>  
Rendered body type: text/plain

>To be honest, I don't really know what "NVIS" means - can anyone help?

"Near-Vertical Incident Skywave" -- It means a short-distance path (that would normally be covered by groundwave) that instead uses ionospheric reflection. There's been a lot of hype recently about "NVIS antennas" which are just antennas designed to minimize ground wave (low-angle) radiation. (i.e. the opposite of what you want for long-distance communication.) The advantage is that for short paths you get less fading, because there is no ground wave path to destructively interfere with the skywave.

AL N1AL

---

## 9. Help.

From: Siddharth Bhattachara <sid@xm.doe.ernet.in>  
Date: 15 Jun 95 10:02:14 IST (+0530)  
To: hfsig@tapr.org  
Message-ID: <A.805216833@xm.doe.ernet.in>  
Rendered body type: text/plain

I have just become a subscriber of this wonderful mailing group and I find a lot of interesting things happening here.

This is my first posting to the list. I need info on the following :

1. The german pactor level-II modem PTC-II's availability. Whether the modem can be purchased in kit form or whether any documentation on its design is available.

2. The PC-PACTOR software to operate Pactor with the AN-93 modem which I intend to build. I could locate the PCTOR and download it from the tapr ftp site. Please let me know where from I can get the PC-PACTOR.

Thanks and cuagn later.

-----  
Siddhartha Bhattacharjee           Internet : sid@mahavir.doe.ernet.in  
Senior Scientific Officer,       Amateur Packet Radio :  
MAEP, C & I Division,           vu2sid@vu2dpg.del.ind.as  
Room No. 3091, Department of Electronics,  
C.G.O. Complex, New Delhi       Phone : 4363102 Ext.3491/4363112  
-----

---

## 10. Re: Help



|        |      |           |              |                         |                |
|--------|------|-----------|--------------|-------------------------|----------------|
| DL1ZAM | DZAM | Martin    | 3.583.25     | Maintal                 |                |
| DL1ZAV | DZAV | Rudolf    | 14.072.5     | Karben or Bangkok/HS    | 2MB-BBS        |
| DL2FAK | DFAK | Tom       | 14.079       | 3.583.25 Hanau          |                |
| DL2GRF | DGRF | Frank     | 14.079       | Offenburg               |                |
| DMEN   | DMEN | Ingo      |              | Oberstaufen             |                |
| DL3MFH | DMFH | Gerd      |              | Muenchen                |                |
| DL3RCF | DRCF | Christian | 14.076       | Poppenricht             |                |
| DL4BCA | DBCA | Reinhard  |              | Bremen                  |                |
| DL5YDJ | DYDJ | Joachim   |              | Boffzen                 |                |
| DL6MAA | DMAA | Peter     |              | Mindelheim              |                |
| DJ7PO  | DJPO | Joerg     | 14.079       | Offenburg               |                |
| DJ9VB  | DJVB | Rolf      | 14.060.8     | Oberhausen              |                |
| DJ0JV  | DJJV | Nussi     |              | Muenchen                |                |
| DJ0OW  | DJOW | Roy       | 14.079       | Offenburg               |                |
| EA5FIN | EFIN | Arthur    | 7.039        | 14.072/74/76.5 La Manga | (PA0LAM)       |
| EA5XK  | EAXK | Herbert   |              | Moraira                 | (DL1VR)        |
| HB9NP  | HBNP | Fred      |              | Mutschellen             |                |
| HB9BDM | HBDM | Chris     |              | Umiken                  |                |
| HB9BIQ | HBIQ | Tad       |              | Niederwil               |                |
| HB9COK | HCOK | Alfred    | 3.579        | 14.060 Mutschellen      | (on weekends)  |
| HB9EBW | HEBW | Werner    |              | Ruenenberg              |                |
| HB9LAR | HLAR | Chris     |              |                         |                |
| JK1DNW | JDNW | Jo        |              | Nagoya/JA               |                |
| KB6BT  | KBBT | Walter    | 14.073/78/79 | 21.073 California       |                |
| KB8LUJ | KLUJ | Phil      | 3.642        | 14.079 Ohio             |                |
| KR8R   | KKRR | Gerhard   | 3.642        | 14.090 Ohio             |                |
| OE9ERI | OERI | Helmut    | 3.583.25     | Rieslern                |                |
| VE3BGE | VBGE | Mike      | 14.077/79    | Toronto                 | 13/14:00       |
| Utc    |      |           |              |                         |                |
| VK6BCP | VBCP | Walter    |              | Perth                   |                |
| W1BEL  | WBEL | Gwyn      |              | Tampa/FL                |                |
| W9UWE  | WUWE | Julius    | 7.035.6/79   | 14.073/79/85            | 21.079 Chicago |
| WA2MFY | WMFY | Pete      |              | (any SYS/2 frequency)   | New Jersey     |
| XE1H0S | XH0S | Henry     |              | Mexico                  |                |

Who is the next on this list...YOU ??

If you know some more Call's send update Info to SysOp 7Z1AB

tnxs 73 de Rudolf HS0/DL1ZAV SysOp @7Z1AB.#APL.SAU.AS  
TY1PS de S92ZM>

## 11. [HFSIG:404] Re: New Clover board

From: Peter Schulze <72253.2602@compuserve.com>  
Date: 16 Jun 95 09:44:27 EDT  
To: TAPR-HFSIG <hfsig@tapr.org>  
Message-ID: <950616134426\_72253.2602\_EHJ113-6@CompuServe.COM>  
Rendered body type: text/plain

Hi everybody,  
Bill Henry gave me one of the new P38 boards in order to make sure that my Express software works ok on it. The board runs well and caused no problems so far. I will post the technical specs tomorrow as i don't have them at hands now here. Amazingly my board does the two fast modes that apply amplitude modulation. Maybe i have an early board, and they cut that off later for commercial reasons. But this shows that the DSP is fast enough for that.  
more tomorrow...

73 Peter TY1PS

## 12. Pactor 2 part 3

From: Peter Schulze <72253.2602@compuserve.com>  
Date: 16 Jun 95 09:44:51 EDT  
To: TAPR-HFSIG <hfsig@tapr.org>  
Message-ID: <950616134450\_72253.2602\_EHJ113-7@CompuServe.COM>  
Rendered body type: text/plain

PACTOR-II

The new Dimension in Data Transmission Technology

By Dr. Tom Rink, DL2FAK, and Hans-Peter Helfert, DL6MAA

## I. Introduction

All new modes should provide significant improvements over existing systems, which must not only refer to the maximum throughput and the robustness. Other basic attributes, like signal bandwidth, required frequency accuracy and compatibility to existing standards also have to be taken into consideration.

Modulation and encoding methods that supply high throughput rates, e.g. 16-DPSK, normally suffer from a lack of robustness. On the other hand, systems distinguished by a high robustness, e.g. DBPSK combined with a rate 1/2 convolutional code, only provide a low maximum throughput. Therefore adaptive digital modes have to be used, that apply different modulation and encoding methods depending on the channel quality. Just changing the symbol rate however, leads to only little adaptivity and additionally results in variations of the bandwidth. In order to prevent any spillover in adjacent channels, the bandwidth should ideally always remain the same, regardless of whether a robust or a fast data transmission is performed. As 500 Hz CW filters are very commonly used and due to the usual spacing of the mailbox frequencies on short wave, a maximum bandwidth of 500 Hz should not be exceeded. In addition, there should not be too high a demand placed on the transceiver used, regarding its frequency adjustment and stability. For optimum results, maximum frequency deviations similar to FSK modes should be tolerated. This forces the use of powerful tracking methods, which allow a link also to be established if the deviation is up to  $\pm 80$  Hz. Further, a new mode should be backwards compatible to an existing standard, preferably with automatic switching, in order to prevent a deficiency of QSO partners in the early stage. PACTOR-II meets all the above mentioned requirements. It is fully backwards compatible with the current PACTOR standard, as the initial link setup is still done in FSK. If both stations are capable of Level II, an automatic switching is performed. The PACTOR-II protocol basically uses a two-tone DPSK system with raised cosine pulse shaping, which reduces the required bandwidth to less than 500 Hz. The maximum absolute transfer rate is 800 bits per second. Due to the improved on-line data compression, maximum effective throughput rates of more than 1200 bits per second can be obtained. PACTOR-II is thus the fastest short wave digital mode. Very efficient error control coding using a convolutional code with a constraint length of 9 and a real Viterbi decoder with soft decision is applied at all speed levels, in addition to analog Memory-ARQ. PACTOR-II is also therefore, by far the most robust digital mode, which allows a link to be established and achieve a reasonable throughput in such poor propagation conditions that all other modes fail. In comparison with the current FSK PACTOR standard including analog Memory-ARQ, which had been the most robust digital mode until the release of PACTOR-II, a further gain of robustness of around 7 dB could be obtained. The following chapters describe some details of the PACTOR-II protocol.

## II. Structure and Timing of the PACTOR-II Frames

Similar to the current FSK PACTOR standard, PACTOR-II is also a half-duplex synchronous ARQ system without any mark/space convention. It may thus be operated in USB as well as LSB position of the transceiver. The initial link setup is still performed using the FSK (PACTOR-I) protocol, in order to achieve compatibility to the previous level. If both stations are capable of PACTOR-II, an automatic switching to the higher level is performed. The basic PACTOR-II frame structure is similar to PACTOR-I. It consists of a header, a variable data field, the status byte and the CRC. The standard cycle duration does not differ from FSK PACTOR and is still 1.25 seconds, which is one of the requirements to obtain easy compatibility to Level I. Longer Control Signals (CS) had to be applied to achieve a higher robustness for the acknowledgment signal, required due to the greater robustness of the data channel. The entire length of the standard packet had to be shortened to 0.8 seconds in order not to shorten the maximum possible propagation delay, which is thus still 170 milliseconds. The requirements to operate PACTOR-II regarding the transmit delay and the receiver recovery time of the used equipment therefore remain unchanged in comparison with Level I.

Due to the signal propagation delay, and equipment switching delays, PACTOR-II as well as PACTOR-I has in the standard mode a maximum range for ARQ contacts of around 20,000 km. As with PACTOR-I, a long path option is available also for PACTOR-II, enabling contacts up to 40,000 km. The sending station calls the partner station in 'Long Path Mode'. Initial contact is established using the PACTOR-I FSK protocol as previously mentioned, but with a cycle time of 1.4 seconds instead of 1.25. This longer cycle time allows for the much greater propagation delays found on 'Long Path' contacts. The link then automatically switches to PACTOR-II, with the same cycle duration. In the new data mode (see below), timing is also automatically adjusted to obtain longer receiving gaps.

Unlike the previous level, PACTOR-II additionally switches to longer packets if the data blocks are not filled up with idles, (i.e. if the transmitter buffer indicates that more information has to be transferred than fitting into the standard packets). If the information sending station (ISS) prefers to use long packets, it sets the long cycle flag in the status word. The information receiving station (IRS) then finally can accept the proposed change of the cycle duration by sending a CS6. This situation, for example, occurs when reading longer files out of mailboxes. The long packets are basically made up like the short ones, but consist of a larger data field, which may contain up to 2208 bits of usable information. The length of these data packets is 3.28 seconds, which leads to an entire cycle duration of 3.75 seconds in this so-called data mode.

When entering text manually in QSO traffic from operator to operator, the maximum throughput of the standard mode is normally not used up, thus the required higher flexibility of the system is still available due to the short packets. The efficient use of longer data packets on short wave is generally only possible, if powerful error control coding, with full frame interleaving is applied to cancel out error bursts or short fading periods. As already mentioned, PACTOR-II uses a convolutional code with a constraint length of 9, a real Viterbi decoder and soft decision, in addition to analog Memory-ARQ. Due to the high coding gain and the resulting capacity of error correction without requesting a repetition of the entire packet, a significant increase in the effective throughput could be obtained. Proceeding from average bit error rates on short wave channels, simple block codes are usually unable to provide enough coding gain. This often leads to a decrease in speed when using longer data strings, as repetitions often cannot be avoided. For more details on these technical foundations, see the first part of this series.

PACTOR-II uses six different CS, each consisting of 40 bits, all having exactly the maximum possible mutual hamming distance of 24 bits to each other. They thus reach exactly the Plotkin boundary and also represent a perfect code. This allows the advantageous use of the Cross Correlation method for decoding, which is also a kind of soft decision, leading to the correct detection of even inaudible CS. This checking is not only confined to a simple binary principle. A complex analog test procedure is applied, using the fine detail data from the DSP, to evaluate the single CS received, as well as the information summed up in the Memory-ARQ buffer. Similar to Level I, CS1 and CS2 are used to acknowledge/request packets and CS3 forces a break-in. CS4 and CS5 handle the speed changes, and CS6 is a toggle for the packet length. All CS are always sent in DBPSK in order to obtain a maximum of robustness.

### III. Speed Levels and Error Control Coding

As mentioned in the introduction, PACTOR-II uses a two-tone DPSK modulation system. Due to the raised cosine pulse shaping, the maximum required bandwidth is only around 450 Hz at minus 50 dB. ASK, which was also tested in the early stage, provided poorer results in weak conditions compared with a higher DPSK modulation, as different amplitude levels are more difficult to distinguish in noisy channels than more phase levels. Additionally, ASK increases the Crest Factor of the signal. For these reasons, it is not used in the final PACTOR-II protocol. Basic information on these items can also be found in the first part of this series.

PACTOR-II uses instead, different DPSK modulation schemes and various code rates. The Crest Factor of the PACTOR-II signal is therefore only 1.45. The basic code used is an optimum rate 1/2 convolutional code with a constraint length of 9. Codes with higher rates, e.g. rate 2/3 and rate 7/8, can be derived from that code by so-called puncturing. Prior to the transmission, certain of the symbols of the rate 1/2 encoded stream are 'punctured' or deleted, and not transmitted. At the receiving end, the punctured encoded bits are replaced with 'null' symbols prior to decoding with the rate 1/2 decoder. The decoder treats these null symbols neither as a received '1' nor as '0', but as an exactly intermediate value. No information is thus conveyed by that symbol that may influence the decoding process. The coding performance of 'punctured' code operation nearly matches the coding performance of the best known classic rate 2/3 or 7/8 codes with a comparable constraint length, provided that the puncture pattern is chosen carefully. The major advantage of this approach is that a single code rate decoder (in our case a rate 1/2 decoder) can implement a wide range of codes.

In the PTC-II, the Viterbi algorithm is implemented for decoding of the convolutional code. Nevertheless, as already indicated in the first part, there are several different methods to decode the PACTOR-II signal, which require less processing power, but in return for this, also provide less coding gain. However, these methods at least allow compatibility to the PACTOR-II standard and may therefore be applied in cheaper hardware.

The most robust PACTOR-II speed level employs DBPSK with rate 1/2 coding, which per cycle allows an absolute throughput of 5 bytes in the standard mode and 36

bytes in the data mode respectively. In the next step, DQPSK with rate 1/2 coding is applied, which leads to an absolute throughput of 14 bytes in the standard mode and 76 bytes in the data mode. This is followed by 8-DPSK with a rate 2/3 coding, providing a throughput of 32 and 156 bytes per packet, respectively. Finally, in best propagation conditions, PACTOR-II applies 16-DPSK with a rate 7/8 coding, which allows the maximum throughput of 59 bytes in a short packet and 276 bytes in the data mode. The mentioned transfer rates are all net rates referring to 8-bit ASCII, which are calculated after the error control coding and all other protocol overhead. As data compression is usually active, these throughput rates must be multiplied by the compression factor. The effective speed is therefore considerably higher in practice. The speed levels are automatically chosen by the PTC-II, considering the link statistics and the actual channel quality, thus no user intervention is required.

#### IV. On-line Data Compression

Like in the previous FSK PACTOR system, automatic on-line Huffman data compression is applied. Additionally, PACTOR-II uses run-length encoding and, as a further novelty, Pseudo-Markov Compression (PMC, see below). Compared to 8-bit ASCII (plain text) PMC yields a compression factor of around 1.9, which leads to an effective speed of about 600 bits per second in average propagation conditions in data mode. PACTOR-II is already around 3 times faster than PACTOR-I and 15 times faster than AMTOR on average channels. However, the maximum effective speed in good conditions can exceed 1200 bits per second. As the PTC-II firmware automatically checks, whether PMC, Huffman encoding or the original ASCII code is the best choice, which depends on the probability of occurrence of the characters, there is no risk of losing throughput capacity. PACTOR-II is of course still able to transfer any given binary information, e.g. programs or picture- and voice files. In these cases the on-line data compression is automatically switched off.

Ordinary Huffman compression exploits the 'one-dimensional' probability distribution of the characters in plain texts. The more frequently a character occurs, the shorter has to be the Huffman symbol that is assigned to the actual character. On the other hand, Markov compression can be considered as a 'double' Huffman compression, since it not only makes use of the simple probability distribution, but of the 'two-dimensional' probability. For each preceding character, a probability distribution of the very next character can be calculated. For example, if the actual character is 'e', it is very likely that 'i' or 's' occurs next, but extremely unlikely that an 'X' follows. The resulting probability distributions are much sharper than the simple one-dimensional distribution and thus lead to a considerably better compression. Unfortunately, there are two drawbacks: Since for each ASCII character a separate coding table is required, the entire Markov coding table becomes impractically large. Additionally, the two-dimensional distribution and thus also the achievable compression factor depends much more on the kind of text than the simple character distribution.

We have therefore chosen a slightly modified approach which we called Pseudo-Markov Compression, because it can be considered as a hybrid between Markov- and Huffman encoding. In this variant, the Markov encoding is limited to the 16 most frequent 'preceding' characters. All other characters trigger normal Huffman compression of the very next character. This reduces the Markov coding table to a reasonable size and also makes the character probabilities less critical, since especially the less frequent characters tend to have unstable probability distributions. Nevertheless, for optimum compression, two different tables for English and German texts are defined in the PACTOR-II protocol and automatically chosen by the PTC-II.

#### V. Some Practical Aspects

Similar to Level I, the tones of the PACTOR-II signal are spaced at 200 Hz. Their frequency may be defined freely in steps of 1 Hz by software command, as long as the shift remains 200 Hz. Thus you can easily switch between high- and low-tones, and also adjust any additionally required tone pair. This allows the utilizing of narrow CW filters in all transceivers that provide the option of activating the corresponding filters in the SSB mode.

In the PACTOR-II system, the transferred information is swapped from one channel (tone) to the other in every cycle. Unlike FSK systems, the link is thus not blocked when strong narrow band QRM completely overpowers one channel (e.g. CW or carriers), but only its maximum speed is reduced. Usual FSK systems with a mark/space convention and without Memory-ARQ have to fail in such cases, because even if a so-called 'space-only' mode is applied, the strongest signal is automatically chosen. This always leads to a break-down of the link, as the QRM is stronger than the useful signal in the proposed case.

PACTOR-II provides a comprehensive Listen-Mode, which is much more robust than known from PACTOR-I, because just the short header has to be received correctly, then the powerful error control coding can be fully utilized. Burst errors may be corrected also by monitoring stations and thus virtually do not affect the performance. The Unproto-Mode in PACTOR-II allows to choose between all the



above mentioned speed and encoding levels. On the receiving side, the correct mode is detected automatically and therefore needs no user-adjustment. For example, a fast and very robust QTC mode can thus be achieved, when a message is transmitted in the Unproto-Mode using DBPSK with rate 1/2 coding.

### 13. pactor 2 part 2

From: Peter Schulze <72253.2602@compuserve.com>  
Date: 16 Jun 95 09:45:14 EDT  
To: TAPR-HFSIG <hfsig@tapr.org>  
Message-ID: <950616134513\_72253.2602\_EHJ113-8@CompuServe.COM>  
Rendered body type: text/plain

This article was originally published in the ADRS Digital Journal.

#### PACTOR-II

The new Dimension in Data Transmission Technology

By Dr. Tom Rink, DL2FAK, and Dipl.-Ing. Martin Clas, DL1ZAM

#### Part 2: Description of the PTC-II

##### I. Introduction

This second part of the PACTOR-II series explains the final version of the PTC-II hardware. As already indicated in the first part, the DPSK modulation system makes it mandatory to use a DSP as the interface to the short wave transceiver. Additionally, the convolutional coding with Viterbi decoder requires very high processing power, more than available in any existing modem on the Amateur Radio market.

However, the many wishes, ideas and suggestions from the ranks of the Radio Amateurs, have finally led to the development of an even more complex and flexible unit than necessary to implement the PACTOR-II system. For example, it has been the wish of many for a long time, to be able to operate not only with usual digital HF modes, but also to cover VHF and UHF Packet Radio at the same time, with one unit. Unlike currently available in existing units, a comfortable built-in mailbox should allow simultaneous access from all communications channels with the same priority, for example in Packet Radio on 23 cms with 9600 baud and on 70 cms with 1200 baud. Naturally, during this time no PACTOR or AMTOR connect on HF should be lost or ignored. This forces the use of a RISC (Reduced Instruction Set) processor for fast processing of the HDLC Packet protocols. As the majority of modern HF transceivers can be programmed via a serial interface, another wish that has slowly formed, is to use this feature for remote control of the transceiver. To set up a mailbox that scans various frequencies on HF, external computer and software is no longer required. Additionally, you could utilize your home based PTC-II whilst on the move, or from a remote location using Packet Radio. Instruct the HF transceiver to tune to a specific frequency, then to make a connect in PACTOR to a particular mailbox and read the messages.

These and many more ideas have led to hardware of very high performance. Due to the use of signal processing, the PTC-II is very flexible, and allows a great variety of different applications to be implemented. These range far wider than those written of up to now. Those technically talented users can make their own programs and modules, which may be loaded via the serial interface into the flash memory of the PTC-II. The possible uses are almost unlimited, as the PTC-II provides a complete, very high performance development platform for DSP and 68020 applications. From simple external control functions (e. g. automatically turning the antenna to the connecting station), or audio processing functions (e. g. super-steep sided FIR filters 400th order for CW, automatic N-times notch filters, etc.) right up to complex functions, like an audio spectrum analyser or extra operating modes, whether known or still to be invented, may be implemented.

The PTC-II comes on the market in February 1995. However, as some of the components used, among them the new DSP chip, are still difficult to get and due to the great demand for the PTC-II, the time of delivery for a unit will initially be about 6 weeks.

##### II. The Processor Section

As has been made clear in the introduction, to achieve simultaneous processing of three communications channels along with the signal coding of PACTOR-II, a powerful computer is essential. A 32-bit processor system has been designed, based on the new communications processor 68360 (QUICC) from Motorola. This processor contains an expanded 32-bit version of the well known 68020 CPU, as used in many powerful computers, together with four separate programmable serial communications ports, the so-called SCC's, implemented as a RISC processor. Two of these SCC's serve as the interfaces to the Packet Radio modems. The third SCC is used as RS-232 interface to the terminal. A buffer chip

MAX207A provides the correct RS-232 voltage levels. The baud rate to the terminal is detected automatically, but can also be determined by software command. The last remaining SCC serves as interface to the HF transceiver for remote control purposes. Four RAM chips with 8 bits each are required, to cover the 32 bit wide data bus. These can vary between 4 times 32k x 8 up to 4 times 512k x 8. The PTC-II can therefore have a maximum of 2 MB of static RAM, which plays a large part in running the mailbox and internal administration as well as external programs, that may be loaded into the PTC-II in addition to the operating software. In order to further expand the possible applications, the PTC-II may also contain additional dynamic RAM in the form of a 72 pin SIMM module of up to 32 MB. The operating system is to be found in a flash memory of up to 512k x 8. Additionally, it is possible to load programs over the serial interface from the terminal, which would enable the PTC-II to do a totally different job, as already mentioned in the introduction. Operating parameters for the PTC-II that should be resistant even to a deep reset are also stored in the flash memory. Data in this kind of memory remains stored even when no voltage is applied, but contrary to an EPROM, it may be electrically erased and re-written whilst in circuit. That makes it very easy updating the system or running different programs on the PTC-II. A battery-backed-up real-time clock and other features of the previous PTC are of course still included.

### III. The HF Modem with Signal-Processor

The 50 MHz version of the XC56156 DSP from Motorola forms the interface to the HF transceiver. As the clock frequency is programmable, it is automatically adjusted to suit the work of the moment. For easy tasks, such as FSK, the processing speed can be reduced, yielding a corresponding saving in energy. The DSP contains a built-in 16-bit digital to analog converter, with the help of which the audio output signal to the transceiver is generated, be it simple AFSK or the complex phase modulation of PACTOR-II. The output amplitude is also programmable and may be set in the range between 10 and 4000 mV by software command. The normally required 'Mic Gain' potentiometer is thus missing. It is also possible for the PTC-II to control the output power of the transceiver, so that the power to maintain the link may automatically be adjusted to an optimum value. No more power than needed being used, which not only saves on the electricity bills, but can considerably extend the life of transmitting components and additionally causes less interference to other stations. For the signal input, the DSP uses a Sigma/Delta analog to digital converter with a 16-bit dynamic range (14 bit effective), which enables the normally necessary Anti-Aliasing filter to be dispensed with. With the exception of the decoupling OP-AMP at the input and output of the DSP, no further components in the signal path are required. The DSP contains some built-in static RAM, which in the PTC-II, is further expanded with 4 additional, very fast, static RAM's. The size of this RAM is 64k-words (16 bit) and is not variable. This enables difficult algorithms, for example 4096 point FFT, to be used. As the DSP has direct access to the main processor data bus, it does not tie up an SCC. The exact receive frequency of PACTOR-II is very quickly and reliably adjusted by software, using a newly-developed tracking method. In addition, the DSP is able, through pulsing the up/down function (which almost every modern transceiver has as microphone buttons) to automatically change the tuning for optimum results. The up/down keys are simulated by transistors, which pull the respective connection to ground.

### IV. The PTC-II Power Supply

The PTC-II contains two power supply input options. Either it may be supplied via a special DC input connector, or directly from the HF transceiver via the connecting cable and socket. The two options are decoupled via diodes and feed a switching regulator. This has a high efficiency, and generates the 5V supply for the digital section. The supply voltage can vary between 8 and 20 volts. The current requirement, due to the use of the switching regulator, is dependant upon the supply voltage and the Packet Radio modems used. The higher the supply voltage, the lower the current consumption. This reverse proportionality is due to the fact that the power consumed is a product of voltage and current, and must be virtually the same before and after the regulator. The efficiency of the regulator is almost unaffected by the value of the supply voltage. The power supply input of the PTC-II contains special filtering, so that the switching harmonics from the regulator cannot reach the outside world. The operating voltage is internally fused with a 5x20 mm fuse. Of course an extra fuse is delivered with each unit to prevent problems in countries with different standards.

### V. The Display and Indicator Unit

The display and indicator unit is built on a separate circuit board, and sits at right angles to the main board, connected by soldering pads. It carries a tuning indicator of 15 LED's, 15 further LED's to display the various operating parameters and a 10-character 5x5 dot matrix LED display. Most of the LED's, including the tuning indicator, are dual colour types, to increase the informa-

tion density and the ease of reading the display. The tuning indicator, for example, changes from red to green as soon as the tuning is optimum for the chosen operating mode. The 10-character LED display shows the operating mode and thus eases the display of any possible future update modes, as the PTC-II is more than sufficient for modes such as FAX, CLOVER-II, etc. Additionally, various status information as well as the call of connecting stations is also given on this display, thus in many cases there will be no need to switch on a terminal. The display is readable from a distance and from unsuitable viewing angles. The brightness is programmable and can be adjusted by a software command.

#### VI. The Packet Radio Modems

The ability to operate Packet Radio with the PTC-II is an integral part of the operating system, and therefore available on all units. The PTC-II is, however, mainly an HF controller, and thus the Packet Radio modems are implemented as modules, to be plugged in if the need arises. This enables a certain amount of mechanical compactness (the entire PTC-II is approximately the size of a modern VHF mobile transceiver). It also prevents those who only need an HF system from being forced to pay for an unwanted Packet Radio modem. The PTC-II contains connectors in the form of double PCB strip headers, on to which the modems can be easily plugged as required. There is space provided for two modems, which are automatically sensed when present. Two types of Packet Radio modems will be offered. A simple and cheaper version using the well-known modem chip TCM3105 for 1200/2400 baud, as well as a version with the XC56156 DSP chip. The DSP version is able to accommodate all baud rates from 300 to 9600 baud. The switching is accomplished by a software command. DSP programming and the clock are supplied from the main board, so that, apart from the DSP chip itself, virtually no extra components are required for the signal processing. Additionally, the DSP modem board has space for an EPROM and its own clock generator, as well as baud rate switching. These enable the modem board also to be used either as a stand-alone modem, or together with other equipment. It delivers the usual Packet Radio signals of Rx/D, RxClock, Tx/D, TxClock and DCD. It is thus compatible with all other Packet Radio systems. Even the connections to the double PCB strip header are compatible with the usual Packet Radio system standards.

The Packet Radio boards are not initially available however. They will come on the market in the middle of 1995, together with the corresponding firmware upgrade for the PTC-II.

#### VII. The PTC-II Construction

The PTC-II is made up of two printed boards, a main board of 147x170 mm, and a front board which, as described above, contains the displays and is mounted at right angles to the main board. The main board is a 6-level multi-layer construction and contains internal ground and supply voltage areas. On the back is the DC input connector, an on/off switch, an 8 pin DIN connector for the HF transceiver, two 5 pin DIN connectors for Packet Radio, an 8 pin Mini-DIN socket for the transceiver control as well as a 9 pin SUB-D socket for the terminal connector. All DIN plugs are delivered together with the unit. Every single pin of every socket has its own filter in order to improve the HF rejection in strong RF fields, as well as to prevent unwanted radiation of electromagnetic energy. On the front is a row of 52 finger-pads for solder connection with the front board, a mounting method used on all SCS controllers. The construction is largely SMD. The flash memory and four static RAM's are socketed to enable easy system upgrades. The RS-232 interface chip is also in a socket to allow easy replacement in event of damage. The SIMM module, due to its construction, needs a holder without which it cannot be mounted. An 8 way DIL switch is also included so that various parameters may be set that should not be changed via software. The whole is enclosed in an aluminium profile case, well known from previous SCS-PTC's and many TNC's. Both front and rear of the case are silk-screen printed, at the front using three colours. The green and red lettering of the dual colour LED's explain their meaning when lighting green or red, respectively.

---

## 14. PACTOR-2 part 1

From: Peter Schulze <72253.2602@compuserve.com>  
Date: 16 Jun 95 09:45:40 EDT  
To: TAPR-HFSIG <hfsig@tapr.org>  
Message-ID: <950616134539\_72253.2602\_EHJ113-9@CompuServe.COM>  
Rendered body type: text/plain

This article was originally published in the ADRS Digital Journal.

### PACTOR-II

The new Dimension in Data Transmission Technology

By Dr. Tom Rink, DL2FAK, and Hans-Peter Helfert, DL6MAA

## I. Introduction

PACTOR was designed more than five years ago in Germany, in order to overcome the known disadvantages of AMTOR and Packet Radio. PACTOR is a cheap and reliable means of fast, robust and error-free data transfer over short wave links, and which does not exceed the usual 500 Hz bandwidth limit for digital modes.

For the first time, not only the complete ASCII character set, but any given binary information could be transferred over short wave, even in very poor propagation conditions. Another aim of the system development was the utilizing of inexpensive and widespread hardware. Since 8-bit controllers without Digital Signal Processor chips (DSP's) were state of the art at that time, Frequency Shift Keying (FSK) had to be chosen as a modulation method. Up to now, PACTOR with analog Memory-ARQ is still the most robust digital mode used in Amateur Radio. It is also still the fastest FSK mode that fits into a 500 Hz channel. These may be some of the reasons that have made PACTOR a standard, which is not only included in virtually all short wave modems in the Amateur Radio market, but is also widespread in the commercial world.

In the meantime however, more powerful CPU's and DSP's have been developed. Processing power that greatly exceeded the financial limits of the average Radio Amateur a few years ago, has dropped to an acceptable price. Some of the current high-end modems for short wave operation already include a DSP, and in a few years you can expect all new modems to contain these chips. The throughput within a 500 Hz channel, as well as the robustness of a system can still be dramatically improved, by using modulation methods different to FSK, combined with powerful error control coding algorithms. A new standard that takes advantage of the forthcoming hardware generation is thus required. These considerations have led to the development of PACTOR-II. This is not 'another new mode', but a fully backwards compatible improvement to the current PACTOR protocol, with automatic switching. As there are already several companies interested in licensing PACTOR-II, the German development group made sure that an implementation in units different from the original PTC-II will also be possible. However, using a less powerful hardware means sacrificing at least a considerable part of the weak signal performance of the system.

This is the first of a series of three parts that describes the PACTOR-II system and the ideas behind it. In this first chapter, we will just explain some important technical fundamentals that everybody has to be aware of, to understand the advantages of the new protocol. The second part describes the PTC-II hardware. The third part deals with details on the PACTOR-II protocol.

## II. Modulation Methods

### 1. Frequency Shift Keying (FSK)

FSK was the first teletype modulation method used, and is still by far the most widespread one. The information is encoded using rectangular pulses, represented by an interleaved ON/OFF keying of two carrier frequencies. The symbol rate is defined as fraction  $1/T$ , the so-called baud rate.  $T$  represents the symbol duration. In FSK systems, the typical wide spectrum that could be expected when modulating a rectangular baseband signal on a carrier, is cancelled out by avoiding phase discontinuities between successive pulses. Thus only the main lobes around the two carrier frequencies are dominant. The amplitude of a 200 baud signal at 500 Hz bandwidth is about 30 dB smaller than the amplitude in the center of the spectrum. If however, the baudrate is increased, the bandwidth of the main lobes will naturally also increase. A 300 Baud signal, for example, clearly exceeds the 500 Hz bandwidth, hence an ordinary CW filter cannot be applied without distorting the pulses. This greatly deteriorates the performance and thus also the Bit Error Rate (BER) of the system. Additionally, signals with higher baud rates suffer from a significant loss of immunity against time smearing (see below). For these reasons, 200 baud is commonly considered to be the maximum useful symbol rate of 2-tone FSK systems, operating over short wave links. Additionally, with regard to the over-crowded frequencies, all new systems should generally require not just a narrow bandwidth, but provide an improved spectral efficiency to obtain a higher throughput. We have therefore to look for a different approach, instead of using FSK.

### 2. Phase Shift Keying (PSK)

In PSK systems, the phase of a signal is used as a means of the information transfer. However, as the HF propagation conditions sometimes change quite rapidly, it is very difficult to track the absolute phase of a signal. Therefore it has proven to be much more efficient to utilize the phase difference between successive pulses. This requires a slightly higher Signal to Noise Ratio (SNR), but in return, the resistance against multipath effects is drama-

tically increased. Modulation that employs phase differences between successive pulses to encode the information is called Differential Phase Shift Keying (DPSK). If there are only two possible phase differences, signalling logical one and zero, the modulation is called Differential Binary Phase Shift Keying (DBPSK). If there are four possible phase differences, signalling logical dibits, it is called Differential Quadrature Phase Shift Keying (DQPSK). For example, with an one-tone 100 baud DQPSK signal, 200 bit/sec. can be transferred. If more phase differences are distinguished, the corresponding systems are called 8-DPSK, 16-DPSK, etc.

As phase shift keying naturally implies phase discontinuities between successive pulses, the spectrum of a DPSK system with hard keying shows the typical strong side-lobes of a rectangular pulse spectrum. The amplitude of a hard-keyed 100 baud DPSK signal is only around 15 dB smaller at a cut-off frequency of  $\pm 250$  Hz than in the center of the spectrum. Thus hard keying must not be used on short wave due to the resulting large bandwidth. To avoid this problem, the baseband signal of a PSK system must be specifically prepared before it gets as far as being modulated on the carrier. This is done by transforming the rectangular pulses containing the binary information into suitable wave forms, using special shaping algorithms. H. Nyquist has designed a pulse with very useful properties, the so-called raised-cosine pulse, which does not produce any spectral spillover. These pulses do not produce any intersymbol interference either, since their amplitude is zero at the sampling time of successive pulses. Thus they can be overlapped without any interference between them, even if the pulses are four times longer than the corresponding rectangular pulses. This is the reason why a very high information density and a good spectral efficiency can be obtained using raised-cosine pulses. Since a raised-cosine DPSK signal with a symbol rate of 100 baud only occupies a bandwidth of around 200 Hz at -45 dB, it is obvious that two of these signals can be placed together into a 500 Hz channel. The resulting system is called two-tone-DPSK. It can robustly transfer 400 bit/sec. within a bandwidth of less than 500 Hz, if DQPSK is applied or up to 600 bit/sec. within the same bandwidth using 8-DPSK.

### III. Robustness

The simplest test of the robustness of a modulation system is the measurement of its behaviour in presence of Arbitrary White Gaussian Noise (AWGN). DQPSK is known to be more robust than FSK, though it also has a better spectral efficiency. For example, to obtain a BER of  $10^{-4}$ , the required SNR per bit is 10.7 dB when using DQPSK, but 12.3 dB when using FSK. DBPSK requires an even smaller SNR of 9.3 dB in that case, and is thus the most robust mode mentioned. It is also important to remember that signals with many levels, e. g. 16-DPSK, require more energy per bit than DBPSK or DQPSK. Generally, a compromise has to be found between the symbol duration (i. e. the baud rate) and the number of bits that each symbol has to carry. Short symbols do require a greater bandwidth, but a high throughput can be achieved with only a few levels per symbol. As a result, the signal is more robust against AWGN than a system with the same throughput using longer symbols and more levels. On the other hand, short symbols are very susceptible to time smearing (see below) and require a higher bandwidth. DQPSK with 100 baud has proven to be a very good compromise between robustness against AWGN and time dispersion, especially if it is combined with powerful error control coding.

Another very important point for a short wave communications system is the resistance against multipath effects, which occur if there is more than one path between transmitter and receiver. Due to the various delays at the receiving end, the combination of the different received signals does not result in a copy of the original signal, but in a more or less distorted wave form. Three major multipath effects can be distinguished: time dispersion or 'time smearing', frequency dispersion and selective fading. These three effects are closely related to each other. They are strong if the frequency used is much below the maximum usable frequency, and if the distance is long. A single hop path on the 20m band, for example, normally does not suffer from severe multipath effects. However, a DX link on the 80m band at night often provides considerably strong multipath problems. The short term time jitter has a magnitude of up to 5 msec. Larger time smearing can only be observed under very special conditions of the ionosphere. A baud rate of 100 symbols per second has proven to be low enough for almost all possible propagation conditions, especially if powerful error control coding is applied.

Frequency dispersion means that the frequency of the original signal is shifted on the path between transmitter and receiver. It is the same effect as the so-called Doppler shift, which can be observed on signals from low orbit satellites. The magnitude of the Doppler shift on normal short wave paths is only a few Hertz, thus it does not influence ordinary FSK systems. However, the demodulator of a PSK signal needs a very accurate information on the carrier frequency in order to work properly. A DQPSK system with a symbol rate of 100 bit/sec. can only deliver a correct output, if the frequency error is less than  $\pm 12.5$  Hz. Automatic frequency tracking must therefore be applied, which can easily be done on a DSP. The PACTOR-II signal, for example, can automatically be

tracked by the PTC-II within an offset range of  $\pm 100$  Hz. Longer symbols and more levels of a DPSK signal require a much higher frequency accuracy. For example, a 32 baud 16-DPSK signal, as used in CLOVER-II, needs an accuracy of better than 1Hz. Thus even small Doppler shifts deteriorate the demodulation process, because it is not possible to track the frequency fast enough at such a high accuracy.

Selective fading, the third multipath effect, mainly influences FSK systems, as the channel with lower SNR determines the BER of the whole system, if the converter cannot switch to space-only mode. The symmetrical property of a binary FSK signal is destroyed by selective fading. PSK modulation is quite robust against this effect.

#### IV. Intermodulation Products and Crest Factor

Whenever a signal, consisting of two or more parallel carriers or 'tones', has to pass through a non-linear stage, intermodulation products are generated. Special attention has to be paid to the third order products, because they are located relatively close to the original signal components. The final RF power stage(s) of the average Amateur Radio transmitter, are capable of a third order intermodulation performance of about -30dB. A two-tone signal with a shift of 200Hz thus produces third order intermodulation products that are located virtually within the original bandwidth of the signal. This means that there will not be any interference on adjacent channels due to intermodulation effects. However, a signal consisting of four tones that are spaced at 125Hz, will be broadened to around 1100Hz of bandwidth at -30dB when passing through the same stage.

Another item that has to be considered is the Crest Factor, which is defined as the ratio of maximum signal amplitude to average signal amplitude. Modulation systems designed for radio frequencies should always have a low Crest Factor, so that the peaks of the signal wave form do not over-drive the final RF stages. Among other considerations, the Crest Factor is influenced by the number of tones used by a system. The more tones that are used, the more difficult it is to get a low Crest Factor. It is in addition, also influenced by the modulation method. PSK, for example, leads to a lower Crest Factor than Amplitude Shift Keying (ASK).

#### V. Error Control Coding

The use of a reliable modulation system is only one of the essential steps towards the goal of optimum data transmission over the difficult paths encountered on HF radio. Dramatic improvements can additionally be obtained by correct preparation of the data before it gets as far as being transmitted by the modem. To be effective, this process, known as Error Control Coding (ECC), imposes very high computing demands on the system processor. Actually the final limit of achievable transmission reliability depends solely on the processing power used for the ECC. The more power available, the closer the theoretical throughput limit, the so-called Shannon Boundary, can be approached.

The basic idea behind this coding is quite simple: A certain number of redundant bits is appended to the original information that has to be transmitted through a noisy channel. The redundant bits are generated from the original data by applying special rules, depending on the chosen code. Data and redundancy then form a new string of bits, which is called a code word. The ratio of the number of information bits and the whole length of the code word is called code rate. The number of valid code words is obviously less than the number of possible code words. A good code and the corresponding encoding process must produce only those valid code words which have a maximum mutual hamming distance, that means a maximum number of different bits. This maximum mutual distance then allows code words to be recognized and distinguished, even in the presence of received errors. For example, if the valid code words of a specific code have a minimum mutual hamming distance of three, each code word containing a single error can be corrected, as the only valid code word with the greatest similarity then represents the correct one.

Two main approaches of ECC can be distinguished: block codes and convolutional codes. Both always require data interleaving to be effective on channels with burst errors. When applying block codes, the message or packet is divided into short blocks containing only a few bits. Each short block is then encoded separately and forms a relatively short code word. Popular block codes are the Golay code, the Hamming codes and the Reed-Solomon code. Block codes can easily be implemented as they often show a cyclic property and thus do not require much processing power. However, they have proven to be relatively weak, especially if the BER is high. They are only able to correct a few errors in each code word and thus do not provide any benefits in very noisy channels or poor propagation conditions. Additionally, it is very difficult to utilize soft decision when using block codes. Soft decision means that the decoder does not only use binary decisions for the error finding process, but also the analog values provided from the demodulator section. It works similar to the analog

Memory-ARQ of PACTOR and requires an ADC or DSP. If convolutional codes are applied, the entire message or packet is encoded and the resulting code words are longer than the original packet. These codes are very powerful, and their efficiency is only limited by processing speed. The complexity of a convolutional code mainly depends on the length of the tapped shift registers, which work as binary convolver and represent the heart of the convolutional encoder. This specific number is called constraint length. It provides an upper boundary of the coding gain that can be achieved by a convolutional code. Several methods have been developed for the decoding process of these codes. The optimum decoder, which allows maximum likelihood decoding, is called the Viterbi-Decoder. Unfortunately, the relationship between constraint length and complexity of a Viterbi-Decoder is not a linear one, but it increases exponentially. Real-time applications of Viterbi-Decoders were limited to quite short constraint lengths for a long time due to slow processor speeds. Nowadays, using the new generation of DSP's, it has become possible to apply constraint length 9 or even higher convolutional codes. As a major advantage of convolutional codes and the Viterbi-Decoder, soft decision may be implemented and only slightly increases the complexity of the system. PACTOR-II, as implemented in the original German PTC-II, applies a convolutional code with constraint length 9 and soft decision.

---

## 15. Re: [HFSIG:408] Help.

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 16 Jun 1995 09:15:45 -0800  
To: hfsig@tapr.org  
Message-ID: <9506161705.AA30504@ucs.orst.edu>  
Rendered body type: text/plain

Dear Siddhartha and all the recent subscribers,

>  
>I have just become a subscriber of this wonderful mailing  
>group and I find a lot of interesting things happening here.  
>  
>This is my first posting to the list. I need info on the  
>following :  
>  
>1. The german pactor level-II modem PTC-II's availability.  
>Whether the modem can be purchased in kit form or whether  
>any documentation on its design is available.  
>

I think Peter answered that one.

>2. The PC-PACTOR software to operate Pactor with the AN-93  
>modem which I intend to build. I could locate the PCTOR  
>and download it from the tapr ftp site. Please let me know  
>where from I can get the PC-PACTOR.

PC-PACTOR --  
PC-PACTOR for the AN-93 is on ftp.tapr.org /tapr/SIG/hfsig/upload. I have not really finished the program (ARQ part is incomplete) as at the time I missed some crucial details that the Pactor folks (SCS) were reluctant to share. However, I did get help from someone else that I used to complete the PSA soundcard version. I will finish the PC-PACTOR version when I get a break. However, you could certainly use the AN-93 with Tom Sailer's fine program that will give you all the modes, including Pactor. It is also on the same ftp site. Let us know how it works for you.

EXTENDED WELCOME --  
It appears that we have a number of new subscribers after Dayton. Welcome to all of you.

PACTOR II --  
Thanks Peter for the detailed overview of Pactor II.

I had many gripes about the Pactor I implementation, especially an inappropriate frame synch method and the inability for automatic link re-establishment after failure. I did note, however, that some of these weaknesses have been addressed in Pactor II. Maybe this is a winner? Hope so.

Does anyone perhaps know whether the actual protocol for Pactor II is available? Or do one have to pay the licensing fee to obtain that?



I think we now have a good idea of the basic modulation scheme, signaling and ECC, but without the protocol description, experimenter's still cannot do it themselves as I found that out the hard way when I experimented with my own version of Pactor I.

EVM56002 --  
I will have the package including the HF modems (works with PC-TOR or HB9JNX software) by Monday (6/19/95). Look in the hfsig/upload area for it (EVM56K0.ZIP).  
There will be a bit more materials about this project shortly.

73's and keep up the good work.

--Johan, KC7WW

---

## 16. Re: [HFSIG:414] Re: Help.

From: Greg Jones <gjones@tenet.edu>  
Date: Fri, 16 Jun 1995 13:22:02 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199506161822.NAA09274@Kay-Abernathy.tenet.edu>  
Rendered body type: text/plain

According to Johan Forrer:

> PC-PACTOR --  
> PC-PACTOR for the AN-93 is on ftp.tapr.org /tapr/SIG/hfsig/upload. I have  
> not really  
> finished the program (ARQ part is incomplete) as at the time I missed some  
> crucial details that the Pactor folks (SCS) were reluctant to share.  
> However, I did get help from someone else that I used to complete the PSA  
> soundcard version. I will finish the PC-PACTOR version when I get a break.  
> However, you could certainly use the AN-93 with Tom Sailer's fine program  
> that will give you all the modes, including Pactor. It is also on the same  
> ftp site. Let us know how it works for you.  
>  
>

Good News for all those waiting and already on the list for AN-93 modems.  
The volunteer finally finished the work and we went to the board shop  
Monday with the updated board. 3 weeks for that and then another week or so  
to kit and mail.

Letters were mailed to all the people on the AN-93 waiting list earlier this  
week.

The updated AN-93 modem support FSK and AFSK output and the new design  
allows on-board alignment with a PC program (so no test equipment is  
required).

There will be 100 kits done. I believe we have taken orders for 35+ to date.

Cheers - Greg, WD5IVD

---

## 17. TI DSK and the ISA Bus

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Sat, 17 Jun 1995 08:10:38 +0100  
To: hfsig@tapr.org  
Message-ID: <199506170857.AA14092@felix.dircon.co.uk>  
Rendered body type: text/plain

Hi all,

Has anyone got a circuit to interface a TMS320C50 to a PC ISA bus. As you  
know the TI DSK is pretty unusable for real applications due to the method  
it talks to the P.C. I was thinking that as all the bus is available on the DSK  
it should be possible to mount it onto a P.C prototype card and communicate  
with it using either the DSP parallel ports or via shared memory. Still  
using the  
serial port for downloads and debugging.

I am still struggling with my MIL-STD-188-141A 8ary modem, on the DSK  
and I have two main problems, the communications described above and  
the fact that the TI DSK free assembler is not a macro assembler so it is  
difficult  
to write properly structured code (well for me)! (or to pinch their example FFT  
algorithms!)

On a slightly different tack, the swiss red cross are evaluating pactor II



and are very impressed with it.  
It sounds an ideal platform for an ALE controller, so anyone know how to contact  
the german manufacturers?

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 18. Re: [HFSIG:401] Re: some questions

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sat, 17 Jun 1995 13:19:45 +0200  
To: hfsig@tapr.org  
Message-ID: <199506171119.NAA01920@foxbat.pix.za>  
Rendered body type: text/plain

>Dear Ilkka,  
>  
>To be honest, I don't really know what "NVIS" means - can anyone help?  
>  
>Just for interest sake, you may find it useful to browse the files in  
>ftp.tapr.org /tapr/SIG/hfsig/upload  
>There are quite a bit on HF propagation; as related to it's affects on  
>digital transmissions.  
>  
>You will also find two Pactor programs there, one that I wrote "PSATOR/PSA-  
>PACTOR" for the sound card DSP, and another one by Tom Sailer, HB9JNX, that  
>will work with a number of different types of modems, including something  
>that you can put together on your new EVM56002.  
>  
>I have ported the Alef Null kernel to the EVM56002 and thus allows you  
>to run any of the codes that run on the DSPCARD4. This includes 1200 baud  
>BELL 202 and 9600 baud PAM (G3RUH) style modems. The DSP kernel includes  
>all HDLC and KISS interfaces, so all you need to have is the EVM and  
>one of the flavors of NOS/KA9Q on your PC. It's really neat stuff. For the  
>HF side of things, I have some HF modems for the EVM as well.  
>  
>Using the ported kernel, you will also be able to participate in the  
>experimental high speed HF modem work. There also are a number of  
>excellent noise reduction programs using Wiener, LMS and correlation-  
>based algorithms. The latter is by Pawel (SP9VRC) and is a joy to use on  
>CW.  
>  
>I am working on putting together the schematic for my version of the EVM's  
>radio interface and will post the complete package (with the kind  
>permission of the Alef Null group) on the net. Watch for the announcement if  
>you are interested.  
>  
>Think this answers most of your questions. Feel free to e-mail me directly  
>about the EVM56002.  
>  
>73's  
>  
>--Johan, KC7WW  
>(email: forrerj@ucs.orst.edu)  
>  
  
Hi Johan. Kan nie wag nie....  
  
danie brynard  
  
>

---

## 19. EVM56002 program package

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sun, 18 Jun 1995 10:56:52 -0800  
To: hfsig@tapr.org  
Message-ID: <9506181846.AA15333@ucs.orst.edu>  
Rendered body type: text/plain

Hi all,

I have uploaded evm56k0.zip to ftp.tapr.org /tapr/SIG/hfsig/upload.  
We gratefully acknowledge those that gave their kind permission.

Hope you enjoy it.

73's

--Johan, KC7WW

---

## 20. pactor 2

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>  
Date: Mon, 19 Jun 95 11:54:42 nst  
To: hfsig@tapr.org  
Message-ID: <9505198035.AA803589734@ccsmtp.udc.neweast.ca>  
Rendered body type: text/plain

The other day a Posting was made here about Pactor 2 (3parts). Does anyone know the phone number and email address for the company mentioned but not named in the article?

I am extremely interested in getting more info on the PTC-II, including price. Possibly for a commercial application.  
Also I am looking for the availability and price of the Packet Radio Modules.

Paul Russell  
paulr@neweast.ca

P.S. Its too bad this mailing list eats the address of the mail originator..

---

## 21. Re: [HFSIG:416] TI DSK and the ISA Bus

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 19 Jun 1995 08:33:31 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <2DEF98E2261@frl.orst.edu>  
Rendered body type: text/plain

Hi Charles,

I do not know of an easy solution to put that DSK on the ISA bus. You may want to consider using one of JDR's ready-made prototyping cards. I have used those in the past for building PC-based projects. The cards come in either 8-bit or 16-bit flavors and contains all the ISA-bus glue logic. The remainder of the cards are then perforated to take wire-wrap parts. The 8-bit cards costs some \$40 price range. The 16-bit cards are a little more expensive, but not much.

The prototyping cards are quite useful - I have built (seems like a long time ago) a wire-wrapped dual 68010 co-processor card on one of these cards and used a DMA-based host-coprocessor interface.

A bit more along the HF digital lines: let us know how things are progressing on the parallel modem - what kind of demodulation filtering method are you using?

73's

--Johan

---

## 22. Re: [HFSIG:420] Re: TI DSK and the ISA Bus

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Mon, 19 Jun 1995 18:52:08 +0100  
To: hfsig@tapr.org  
Message-ID: <199506191939.AA28644@felix.dircon.co.uk>  
Rendered body type: text/plain

>A bit more along the HF digital lines: let us know how things are  
>progressing on the parallel modem - what kind of demodulation filtering  
>method are you using?

>  
>73's

>  
>--Johan

>  
>  
>  
Hi Johan,

I am working along the lines that Adrian G4ZHZ used in his piccolo modem, the main problem is implementing an FFT as all the examples I have are done using macros, I am trying to write a simple macro expander to take the TI examples and convert them into something the DSK understands. The other

problem is of course the wretched interface to the DSK, I think it will break my coding skills to implement the modem and expect a software uart to work as well !!

I will investigate one of those prototype cards, also they have bought a TI sun based development system at work with both an assembler and C compiler, I am negotiating access to it as you read this !!!!

I have been busy with a Linux system as well so that has eaten some of my time and I am also QRV on H.F from my car now, I have the auto ATU in the trunk and have a Pro-am 14Mhz vertical on a roof rack that took me ages to ground properly.

I have also been playing with Phil's KA9Q Viterbi code on my Linux system, it works great, next thing is to understand what his macros do !

Regards Charles

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

### 23. Re: [HFSIG:414] Re: Help.

From: Siddhartha Bhattacharjee <sid@doe.ernet.in>  
Date: Mon, 19 Jun 1995 18:16:51 +0530 (GMT+05:30)  
To: hfsig@tapr.org  
Message-ID: <Pine.3.89.9506191855.B24453-0100000@mahavir>  
Rendered body type: text/plain

Johan,

I am sorry I could not locate the file PC-PACTOR in the site.  
I also could not locate the Tom Sailer's software which  
you have referred to in your last mail.

I would appreciate if you could let me have the details.

The price of the PTC-II is little too much. I understand that  
that the sophistication which goes into it is really very  
great. The protocol looks great but it remains to be seen as to  
how does it perform under a given band condition. I was also  
very much impressed by the article on G-TOR published in the  
May 1994 issue of the QEX. Simplified architecture of the modem  
and the price/performance ratio is also required to be looked into  
to make it popular among the hams.

I would request a feedback from one of the PTC-II users ( I do not  
know if there is a subscriber of this mailing list ) to say  
few words about its performance.

Thanks to all the contributors.

Sid, vu2sid in New Delhi, India.

---

### 24. Re: [HFSIG:421] Re: TI DSK and the ISA Bus

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 19 Jun 1995 13:37:15 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <2E4099210E5@frl.orst.edu>  
Rendered body type: text/plain

Hi Charles,

Appears that you are a busy fellow indeed!

Sorry I cannot help much with the C50' macros. Would it be possible to use  
Bison or other parser-generator to do that?

The Linux connection is interesting. I had a go at it about a year ago, but  
gave up after realizing that it was such an incredible moving target. When  
I started following Pawel's work on HF modems, I realized that the whole  
issue of the modem and it's eventual place in a network, needs some thought,  
worthy of exploration. I have had some success with Pawel's modem - BTW,

the way he does it is to use the Leonid kernel calls to communicate with NOS. This way he has a universal software interface. This is a novel idea (at least I think so). The version of JNOS that I played with ran under DOS, but one must realize that folks that build networks don't use DOS any more, anyway you need to keep up with recent software. Then went on to try OS/2 with very mixed results. I decided to remove OS/2 totally from my system after an awful crash (running NOS) that destroyed both FAT's on both my hard drives (there were other reasons too). I then decided to revisit Linux and been quite amazed by how stable it is. The main objective at present is to get a DSP sound card device driver functional that will enable one to download DSP algorithms and allow development of applications that feeds into Linux's networking capabilities. However, as you can imagine, things have gotten quite involved now: Not only is it knowing the DSP and programming DSP applications, but the Linux device drivers and dealing with TCP/IP as well.

I welcome further comments, ideas, and suggestions on this theme.

73's

--Johan, KC7WW

---

## 25. Re: [HFSIG:422] Re: Help.

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 19 Jun 1995 13:45:48 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <2E42D747399@frl.orst.edu>  
Rendered body type: text/plain

Dear Siddhartha,

Try the following:

ftp.tapr.org /tapr/software\_lib/terminal/PCTOR302.ZIP ---- PC-TOR 3.02  
ftp.tapr.org /tapr/SIG/hfsig/upload/jnx\_2.zip --- HB9JNX software.

You may also find this (and perhaps leater versions) on:

ftp.ucsd.edu /hamradio/packet/tcpip/incoming directory, or in the "DSP"  
or "packet" directories. Things move around a bit.

Hope this helps.

--Johan, KC7WW

---

## 26. Re: [HFSIG:423] Re: TI DSK and the ISA Bus

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Mon, 19 Jun 1995 20:52:35 +0100  
To: hfsig@tapr.org  
Message-ID: <199506192139.AA02968@felix.dircon.co.uk>  
Rendered body type: text/plain

>Hi Charles,

>

>Not only is it knowing the DSP and programming DSP applications, but  
>the Linux device drivers and dealing with TCP/IP as well.

>

>I welcome further comments, ideas, and suggestions on this theme.

>

>73's

>

>--Johan, KC7WW

>

Yes I know exactly what you mean,

I have been through that route as well, I had OS/2 running on a partition,  
I have  
got rid of it now.

I have two machines one downstairs in the shack, on which I run DESQview/X  
and a second machine upstairs that runs Linux. I use the 'Xterminal' in the  
shack  
to log into the Linux ssystem over an ethernet, which I am in the process of  
expanding around the house! The people at work think I am nuts.

As far as the macros are concerned, yes they could be done with lex/yacc. Yet  
another thing to learn!

I have started to port all my 'homebrew' software to the Linux environment, its  
an ideal Hams OS and it great to have the source, there are a number of

things I would like to play with like X.400 mail, but I have to concentrate my efforts on one thing at a time.

I would like to get some propagation S/W like ioncap to run under Linux. I have also heard of other Hams using cron tasks to scan radios etc. It seems as if it would be not too difficult to build a sophisticated system using building blocks such as Linux.

I am also trying to learn SASD using Cadre Teamwork at work, it would be nice to get some CASE tools for Linux as well.

Well this ain't really to do with HFSIG but certainly interesting. Things seem to be quite quiet here recently.

I must admit I have learn't a lot listening to this sig then reading up afterwoods, I have actually found a use for what they tried to teach me at university! I have been going through all my basic math books again and also the ones on modulation etc. Thats what it is all about isn't it.

Regards Charles.

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 27. Re: [HFSIG:423] Re: TI DSK and the ISA Bus

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Mon, 19 Jun 1995 21:23:31 +0100  
To: hfsig@tapr.org  
Message-ID: <199506192210.AA03756@felix.dircon.co.uk>  
Rendered body type: text/plain

Hello again Johan,

Well I have been poking about on ftp.ti.com and have found a file called 5xdskspc.exe which is a compressed file containing an audio spectrum analyser for the DSK. Unlike the code shipped with the DSK which outputs via the D/A port this code talks back via the serial port and has a graphical display on the P.C. I haven't tried it yet but it maybe suitable for an FFT and certainly the C++ routines provide info on how to download code to the DSK from the P.C. I have a look at it tomorrow.

Regards Charles (off to my bed!)

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 28. Re: [HFSIG:419] pactor 2

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Tue, 20 Jun 1995 09:32:14 +0200 (MET DST)  
To: hfsig@tapr.org  
Cc: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.950620092202.27837A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Mon, 19 Jun 1995, Paul Russell wrote:

> Date: Mon, 19 Jun 1995 10:06:18 -0500  
> From: Paul Russell <paulr@hagar.udc.neweast.ca>  
> To: hfsig@tapr.org  
> Subject: [HFSIG:419] pactor 2

>  
> The other day a Posting was made here about Pactor 2 (3parts). Does  
> anyone know the phone number and email address for the company  
> mentioned but not named in the article?  
>  
> I am extremely interested in getting more info on the PTC-II,  
> including price. Possibly for a commercial application.  
> Also I am looking for the availability and price of the Packet Radio  
> Modules.  
>  
> Paul Russell  
> paulr@neweast.ca  
>  
> P.S. Its too bad this mailing list eats the address of the mail  
> originator..  
>

The company that sells the PTC-II is:

SCS, Spezielle Communications Systeme GmbH  
Roentgenstrasse 36,  
D-6450 Hanau, Germany.

Phone/FAX: +49 6181 23368

This is also the home address of DL2FAK, Dr. Thomas Rink.

73's Frode

|                                |        |                        |
|--------------------------------|--------|------------------------|
| Frode Weierud                  | Phone  | : 41 22 7674794        |
| CERN, SL                       | Fax    | : 41 22 7679185        |
| CH-1211 Geneva 23, Switzerland | E-mail | : frode@dxcern.cern.ch |

---

## 29. Re: [HFSIG:392] HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl  
Date: Thu, 22 Jun 1995 11:08 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <190754A320218C58@chopin.ifj.edu.pl>  
Rendered body type: text/plain

Hello everybody,

Just to report little progress I have just made: I have coded  
a 12-tone DQPSK modem into the DSPCARD4.  
12 tones, each carries 125 baud, QPSK (4-PSK) => 2 bits/symbol  
=> total data rate = 12\*2\*125 = 3000 bits/second.  
The tones are spaced by 187.5 Hz, the first is 500 Hz, the last  
is 2687.5 Hz. The whole bandwidth is 100-200 Hz (on both sides)  
larger than that. The modem seems to work better than my previous  
multi-tone DBPSK one.

Infact it's up to the user to define how many tones he need  
and where should they be placed - the code is flexible in this respect,  
so one can easily adapt the data rate to the available bandwidth.

The modem's encoder and decoder is based on the 128 point FFT  
at 8000 Hz sampling.

The modem works on my DSPCARD4 with the build-in KISS-TNC  
or in the test mode so you can spy on various internal data.  
In principle it should work too on the EVM56002 with the Johan's  
BIOS.ASM - I am going to try this. My other applications did work  
so I don't expect a problem there.

No FEC yet, but I am thinking of it. First I should make  
some provision for soft decisions, then interleave, then FEC.

The other thing to think about: how to make such FFT decoder  
"tuneable" so it could accept a signal shifted in frequency.  
For the moment the decoder can detect a mis-tune, but is not able  
to correct it.

Pawel

---

## 30. Re: [HFSIG:428] Re: HFSIG activities

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Thu, 22 Jun 1995 08:43:45 PST8PDT

To: hfsig@tapr.org  
Message-ID: <3272701710E@frl.orst.edu>  
Rendered body type: text/plain

Pawel,

Excellent! If you have the foundation of something that has promise, perhaps it is time that we can divide the task up and let a couple of folks work on it. Just off the top of my head I can think of a couple of them:

- 1) Tuning indicator
- 2) Frequency tune-up/correction
- 3) ECC
- 4) Simple KISS-based user-interface instead of NOS so that we can get at testing the modem. The idea is that NOS compatibility be retained for future networking applications.

These are quite general in nature, so you probably still could change things at the very low implementation level and use the higher levels of code.

What do you think - anyone interested in working on some of these topics?

73's  
--Johan, KC7WW

---

## 31. FAQ ?

From: RHIII@delphi.com  
Date: Thu, 22 Jun 1995 14:23:34 -0400 (EDT)  
To: hfsig@tapr.org  
Message-ID: <01HS0D1FMF9K94GYZ@delphi.com>  
Rendered body type: text/plain

Tremendous ideas and excellent writing on this list.  
Is there a FAQ supporting it ?

---Richard/NT2Z

---

## 32. Re: [HFSIG:428] Re: HFSIG activities

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>  
Date: Tue, 27 Jun 95 10:45:43 nst  
To: hfsig@tapr.org  
Message-ID: <9505278042.AA804275155@ccsmtp.udc.neweast.ca>  
Rendered body type: text/plain

>>The other thing to think about: how to make such FFT decoder "tuneable" so it could accept a signal shifted in frequency.  
For the moment the decoder can detect a mis-tune, but is not able to correct it.

Pawel<<

Try a Zero-Padded Larger FFT (Say 512pt)  
 $8000/512 = 15.62\text{Hz/bin}$

Although a zero-padded FFT will not resolve two tones that occupied the same bin in a smaller FFT into their individual components, it will resolve where the peak power point is of a single tone to a more accurate point. Thus by calculating the magnitude on the bins adjacent to the bin expected to carry the data tone, you can choose which one you should interpret as best centred and to use for phase calculation.

So a 512pt has 4 times more bins than a 128pt, allowing you to choose a bin closer to the true location of the received signal. You can thus compensate for radio offset without having to shift frequency with something like a Hilbert transform, although I'm uncertain which would be more efficient in code execution time.

I don't know how much error would occur in choosing to use a more offset bin part way through a packet (in my simulations it was negligible with the angle in the zero-padded FFT the same as in the smaller FFT), but I suspect the radio's frequency offset drift to be VERY slow. Thus choose the offset during the synchronization, and stick with it for the rest of the packet.

Note also that all the channels in the parallel modem would be shifted by the same amount of freq/bins (at the audio level since freq offset is at the RF level). So calculating the peak power point by adding together the magnitudes of

the corresponding bins at each channel over several symbols will average out which way the radio is offset even in the presence of noise.

How are you calculating the phase angle? I have an article on how to do a simplified arctan approx for a TMS32010 if its useful to anyone. I'm certain it could be recoded to any DSP.

Paul Russell  
paulr@neweast.ca

---

### 33. Re: [HFSIG:431] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl  
Date: Tue, 27 Jun 1995 17:28 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <3BE65F09E022085A@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>Try a Zero-Padded Larger FFT (Say 512pt)  
> 8000/512 = 15.62Hz/bin  
>  
>Although a zero-padded FFT will not resolve two tones that occupied  
>the same bin in a smaller FFT into their individual components, it will resolve  
>where the peak power point is of a single tone to a more accurate point. Thus by  
>calculating the magnitude on the bins adjacent to the bin expected to carry the  
>data tone, you can choose which one you should interpret as best centred and to  
>use for phase calculation.

So you select the best tone, and then you may possibly be mis-tuned  
by up to 7.8 Hz - this is still a lot... In term of phase changes:  
7.8 Hz is about 49 radians/second, I transmit 125 symbols per second,  
thus during one symbol the phase would turn by 0.39 radian (22 degrees)  
- this is a lot, almost half way to make a bit error (45 degrees).  
I know I could make even longer FFTs and get more finer steps...  
I agree this is one way to deal with the problem.

I was actually thinking of a way to "interpolate" between the FFT bins.  
If a tone falls say right inbetween bin 7 and 8 which combination  
of the two you need to get the data as if the tone falls just right  
into the bin's center ?  
In other words how can you frequency-shift the frequency-domain data  
by an arbitrary step ? I know how to do it with the time-domain data...

>So a 512pt has 4 times more bins than a 128pt, allowing you to choose a bin  
>closer to the true location of the received signal. You can thus compensate for  
>radio offset without having to shift frequency with something like a hilbert  
>transform, although I'm uncertain which would be more efficient in code  
>execution time.

Hilbert transform can shift the frequency ? This might be something  
I am looking for ?

>I don't know how much error would occur in choosing to use a more offset bin  
>part way through a packet (in my simulations it was negligible with the angle in  
>the zero-padded FFT the same as in the smaller FFT), but I suspect the radio's  
>frequency offset drift to be VERY slow. Thus choose the offset during the  
>synchronization, and stick with it for the rest of the packet.

This is my current "research" direction. I am thinking of making a header  
such that the receiver may quickly acquire the frequency shift and  
the symbol sync. - possibly both at same time. After the header has passed  
only minor adjustments are to be made.

>Note also that all the channels in the parallel modem would be shifted by the  
>same amount of freq/bins (at the audio level since freq offset is at the RF  
>level). So calculating the peak power point by adding together the magnitudes of  
>the corresponding bins at each channel over several symbols will average out  
>which way the radio is offset even in the presence of noise.

There is a little problem that the bands of the tones partially overlap  
or are at least well packed so you may not be able to find a clear peak.  
The could be overcome by sending only every second tone for the header.

>How are you calculating the phase angle? I have an article on how to do a  
>simplified arctan approx for a TMS32010 if its useful to anyone. I'm certain it  
>could be recoded to any DSP.

I don't really compute the phase angle but compare the two successive I/Q  
vectors. If A(a vector) is what I got from the current FFT and B is what  
I got from the previous FFT I compute:  $X=B*A$  and  $Y=B'*A$ .



\* means the vector-dot-product, ' means turning a vector by +90 degrees.  
X and Y (infact a vector) contains the information to find one the di-bit transmitted.

Anyway, I am certainly interested in a fast way to compute the phase from the the X/Y values.

I made more progress with the intersymbol crosstalk: I parametrized the window's shape and by minimizing the total crosstalk in terms of the parameters (I used praxis.zip found in msdos/math on most FTP sites) I found an optimal shape with crosstalk at  $0.6E-4$  (it used to be  $1E-1$ ). The price for much lower crosstalk was the first side-lobe at about -10dB level... but futher side-lobes are below -50dB (measured with a soundcard + FREQ4 software).

Pawel

---

## 34. Re: [HFSIG:429] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl  
Date: Tue, 27 Jun 1995 17:52 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <3F38291EC022085A@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>Excellent! If you have the foundation of something that has promise,  
>perhaps it is time that we can divide the task up and let a couple of folks  
>work on it. Just off the top of my head I can think of a couple of them:

>From my point of view these are the major problems:

1. Tuning correction: 10 Hz mis-tune seems to be real bad, and 20 Hz makes the signal unreadable even for 100 dB S/N :-)  
We probably need to make it automatic at least within a small range.  
For network application, the packet's header should be made for fast auto-retune and symbol-sync.
2. FEC with soft-decisions... how do we judge on the "strength" of the received symbol ?  
Here comes a related issue: if you want to compare the "strength" of symbols coming from different tones, you should normalize them somehow - this is trivial if you use division, but division is rather costly (in terms of clock cycles) for the DSPs.

So an algorithim is needed which least divisions possible.  
I find my radio's (YEASU FT757-GXII) audio characteristic to be very unperfect: 2500 Hz tone is 2-3 times lower in amplitude than the 800 Hz tone (this is just the receiver). Well, this could be "perfect" for voice, but not for data.

Another issue related to "band equalization" is the symbol synchronization. For the moment I am taking the average symbol shift from all 12 tones. But in general you want to keep the proper sync. even if one or more tones get overcome by a strong carrier-type interference. Thus you should weight the shifts from different carriers.

Pawel

---

## 35. Pawel?

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>  
Date: Tue, 27 Jun 95 15:53:30 nst  
To: hfsig@tapr.org  
Message-ID: <9505278042.AA804293635@ccsmtp.udc.neweast.ca>  
Rendered body type: text/plain

Pawel,

I'll send the file about angle calculations, but my mail server is eating your email reply address, please send it. The 13K file is maybe a bit long for posting here - unless several people want it??

I tried asking the tapr.org listserver for the subscribers, but it indicates that all of us are hidden names - Oh well (Guess that helps keep down people using our names for other purposes...).

>>>Here is the current list of non-concealed subscribers:  
>>>Total number of subscribers: 144 (0 shown here)

There are definitely more subscribers than I thought.

Paul  
paulr@neweast.ca

---

### 36. Re: [HFSIG:433] Re: HFSIG activities

From: Adrian Nash <nash@camail.ca.nmp.nokia.com>  
Date: Wed, 28 Jun 1995 10:17:22 +0100  
To: hfsig@tapr.org  
Message-ID: <199506280918.MAA29723@noknic.nokia.com>  
Rendered body type: text/plain

Hi Pawel

I have a few suggestions which have worked in my MFSK modem which I suppose is very similar in some respects.

- >1. Tuning correction: 10 Hz mis-tune seems to be real bad, and 20 Hz
- > makes the signal unreadable even for 100 dB S/N :-)
- > We probably need to make it automatic at least within a small range.
- > For network application, the packet's header should be made for
- > fast auto-retune and symbol-sync.

There is a simple equation which uses the magnitudes (NOT magnitude squared) of adjacent FFT points to calculate frequency error to within the separation of the points. I have used this in my MFSK modem to track frequency error with some success. However, I can't remember the formula off the top of my head! I'll get back to you with this. I think Johan Forrer may be able to help you with this one. He originally suggested the idea to me and I then produced a MathCad simulation which worked, so then I coded it up in my modem and it works fine for tracking.

I use a Hilbert transform coupled to a complex demodulator to shift the audio down to zero-IF in the complex plane. The tracking loop is closed by applying the exact offset in Hz to the NCO algorithm.

- >2. FEC with soft-decisions... how do we judge on the "strength"
- > of the received symbol ?
- > Here comes a related issue: if you want to compare the "strength"
- > of symbols coming from different tones, you should normalize
- > them somehow - this is trivial if you use division, but division
- > is rather costly (in terms of clock cycles) for the DSPs.

For soft decisions, I use the following equation which unfortunately does have a divide in it. However, the divide only needs to be performed once in the case of an MFSK modem or 12 times in your case for the 12 best signals:

$$C = \frac{I^2 (best)}{n \cdot \frac{I}{k}}$$

ie, accumulate the magnitude squared results ( $I^2+Q^2$ ) then find the best (ie greatest) magnitude. Divide one by the other.  $C(n)^2$  is then a good indicator of the "goodness" or confidence of the particular tone, and is fine for soft decision use. On my ADSP2101-based modem, the divide is done in floating point and takes about 30 cycles (about 1.5us on ADSP2115) including conversion back to fixed point.

Regards  
Adrian

---

### 37. Re: [HFSIG:432] Re: HFSIG activities

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Wed, 28 Jun 1995 07:58:21 -0800  
To: hfsig@tapr.org  
Message-ID: <9506281547.AA24218@ucs.orst.edu>  
Rendered body type: text/plain

Pawel,

- >Hilbert transform can shift the frequency ? This might be something
- >I am looking for ?
- >

It may be useful to use a Hilbert transform especially if your demodulator require I/Q paths, i.e. doing complex FFT's. You do need, however, some good low pass filters to go with it. All these computational blocks eats at your machine ticks, however. If you can fit it in, that will be excellent. We had some discussion about this topic under "HF channel simulator".

>I made more progress with the intersymbol crosstalk: I parametrized  
>the window's shape and by minimizing the total crosstalk in terms  
>of the parameters (I used praxis.zip found in msdos/math on most  
>FTP sites) I found an optimal shape with crosstalk at 0.6E-4  
>(it used to be 1E-1). The price for much lower crosstalk was the first  
>side-lobe at about -10dB level... but further side-lobes are below -50dB  
>(measured with a soundcard + FREQ4 software).  
>

For interest sake, could you perhaps tell us a bit more about the FREQ4 software - where to obtain it etc. These analytical methods and tools are useful to know about.

-----

I also had an interesting e-mail from Adrian, G4ZHZ about his progress with his Piccolo (MFSK) project. As you all know, he is using a DSP sound card and he tells me that he has done a lot of work on the DSP kernel (that part of the DSP code that runs in real time, in parallel with other DSP activities - mainly to orchestrate tasks and communicate with the host PC). This was necessary before he could start porting his exsisting Piccolo code to the DSP. These extensions also are quite general, as a matter of fact he built a very nice audio signal processing box using that. This audio processor is intended for use with your HF radio and includes: various filters (narrow CW etc.), denoisers, and a software BFO to resolve slightly, off-tune SSB or CW. One can quite easily see how these different functions plays a role in his modem developments. The sound procesor, though would be a handy piece of code in its own right. I am looking forward testing it out. Good work Adrian!

73's

--Johan

---

## 38. ADDRESSING PROBLEM

From: "Taylor, Gregg" <GTaylor@ecad00.avionics.itt.com>  
Date: Wed, 28 Jun 95 14:41:14 EST  
To: hfsig@tapr.org  
Cc: Henry\_Sonntag\_at\_ITTAVI@CCMAIL.AVIONICS.itt.com  
Message-ID: <9505288043.AA804376746@CCMAIL.AVIONICS>  
Rendered body type: text/plain  
[Empty message body]

---

## 39. Freq anal. with soundcards

From: Peter Schulze <72253.2602@compuserve.com>  
Date: 29 Jun 95 11:27:27 EDT  
To: TAPR-HFSIG <hfsig@tapr.org>  
Message-ID: <950629152726\_72253.2602\_EHJ164-1@CompuServe.COM>  
Rendered body type: text/plain

> For interest sake, could you perhaps tell us a bit more about the FREQ4  
> software - where to obtain it etc. These analytical methods and tools are  
> useful to know about.

There are several freeware programs around that allow spectrum analysis with soundcards :

### 1.- MICFFT

by Craig Walsh (walsh@biovx1.DNET.NASA.GOV)  
This one works on almost any 'compatible' soundcard.  
Craig told me that he is working on a windows version.  
The most current Version is 1.2.  
Micfft can be found in the CompuServe Soundblaster forum (MICFFT.ZIP).  
There was an article by DK4ZC about using this program to analyse HF digital signals in the March issue of the ADRS Digital Journal.  
MicFFTs main limitation is slow execution and a FFT length of max 256 points. On the other hand it is very stable, has nice graphics and works on about every Hardware. MicFFT is freeware.

## 2. FREQ3, FREQ4 and SPECGRAM

by Philip VanBaren  
Internet: phillipv@engin.umich.edu  
Compuserve: 71214,2302  
1845 Lake Lila Dr. Apt. C3  
Ann Arbor, MI 48105

..also Freeware.  
can be found on Compuserve as well, in the same forum (.ZIP).  
Freq3 is an older Version but comes with sourcecode (Borland C++).  
FREQ3 was written for the Pro Audio Spectrum 16. FREQ4 also  
supports the Soundblaster cards.  
Main advantages against MicFFT:  
- FFT Length variabel, max. 2048 points  
- a lot faster (31ms for a 1024 FFT on a 486/33)  
- with PAS cards up to 44KHz at 16 bit Resolution  
- SPECGRAM has an excellent graphical Display on SVGA.  
- lots of commandline options  
- FREQ3 sourcecode avaiable.  
(thanks to Christoph dl4rcg who supplied the FREQ info to me some time ago)

3. Another usefull program is Mathplot  
by Phillip H. Sherrod  
4410 Gerald Place  
Nashville, TN 37205-3806 USA  
615-292-2881 (evenings)  
Internet: 76166.2640@compuserve.com  
(shareware, uncrippled registration \$25, also on the CIS soundblaster forum).  
Mathplot is a mathematical function plotting system which allows you to  
enter functions using ordinary algebraic notation and immediately plot them.  
Cartesian, polar, and parametric functions are supported, and up to four  
functions may be displayed simultaneously. A variety of built-in functions  
are provided. Plotting commands may be executed from command files.

Maybe someone who is on Compuserve as well can download these programs and  
post them here on the TAPR system ?  
Unfortunately i can't do that as i am on a very (!)  
expensive (-2400 baud X25) line,  
where i am charged 15 cents per kilobyte.  
So moving these files out of Cis and up here  
would cost me a little fortune. The Datahighway did not  
arrive in Africa yet(?)....

73 Peter TY1PS

---

## 40. Re: [HFSIG:438] Freq anal. with soundcards

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 30 Jun 1995 06:36:11 -0800  
To: hfsig@tapr.org  
Message-ID: <9506301424.AA26115@ucs.orst.edu>  
Rendered body type: text/plain

Peter,

Thanks for the interesting summary. Time that I explore some of those  
mentioned programs.

For interest sake, I may just add that the sound card FFT scope (on  
hfsig/upload) will also be helpful, so will Tom McDermot's program for the  
DSP-93, that does the same thing. I am just not certain about whether the  
frequency resolution is sufficient.

--Johan, KC7WW

>> For interest sake, could you perhaps tell us a bit more about the FREQ4  
>> software - where to obtain it etc. These analytical methods and tools are  
>> useful to know about.  
>  
>  
>There are several freeware programs around that allow spectrum  
>analysis with soundcards :  
>  
>1.- MICFFT  
>by Craig Walsh (walsh@biovx1.DNET.NASA.GOV)

>This one works on almost any 'compatible' soundcard.  
 >Craig told me that he is working on a windows version.  
 >The most current Version is 1.2.  
 >Micfft can be found in the Compuserve Soundblaster forum (MICFFT.ZIP).  
 >There was an article by DK4ZC about using this program to analyse  
 >HF digital signals in the March issue of the ADRS Digital  
 >Journal.  
 >MicFFTs main limitation is slow execution and a FFT length of max 256  
 >points. On the other hand it is very stable, has nice graphics and works on  
 >about every Hardware. MicFFT is freeware.  
 >  
 >2. FREQ3, FREQ4 and SPECGRAM  
 >  
 >by Philip VanBaren  
 >Internet: phillipv@engin.umich.edu  
 >Compuserve: 71214,2302  
 >1845 Lake Lila Dr. Apt. C3  
 >Ann Arbor, MI 48105  
 >  
 >.also Freeware.  
 >can be found on Compuserve as well, in the same forum (.ZIP).  
 >Freq3 is an older Version but comes with sourcecode (Borland C++).  
 >FREQ3 was written for the Pro Audio Spectrum 16. FREQ4 also  
 >supports the Soundblaster cards.  
 >Main advantages against MicFFT:  
 >- FFT Length variabel, max. 2048 points  
 >- a lot faster (31ms for a 1024 FFT on a 486/33)  
 >- with PAS cards up to 44KHz at 16 bit Resolution  
 >- SPECGRAM has an excellent graphical Display on SVGA.  
 >- lots of commandline options  
 >- FREQ3 sourcecode available.  
 >(thanks to Christoph dl4rcg who supplied the FREQ info to me some time ago)  
 >  
 >3. Another usefull program is Mathplot  
 >by Phillip H. Sherrod  
 >4410 Gerald Place  
 >Nashville, TN 37205-3806 USA  
 >615-292-2881 (evenings)  
 >Internet: 76166.2640@compuserve.com  
 >(shareware, uncrippled registration \$25, also on the CIS soundblaster forum).  
 >Mathplot is a mathematical function plotting system which allows you to  
 >enter functions using ordinary algebraic notation and immediately plot them.  
 >Cartesian, polar, and parametric functions are supported, and up to four  
 >functions may be displayed simultaneously. A variety of built-in functions  
 >are provided. Plotting commands may be executed from command files.  
 >  
 >Maybe someone who is on Compuserve as well can download these programs and  
 >post them here on the TAPR system ?  
 >Unfortunately i can't do that as i am on a very (!)  
 >expensive (-2400 baud X25) line,  
 >where i am charged 15 cents per kilobyte.  
 >So moving these files out of Cis and up here  
 >would cost me a little fortune. The Datahighway did not  
 >arrive in Africa yet(?)....  
 >  
 >73 Peter TY1PS  
 >  
 >  
 >

# July 1995

Original source file: hfsig.9507

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-07/TAPR-HF\\_1995-07.txt](https://archive.org/download/TAPR-HF_1995-07/TAPR-HF_1995-07.txt)

## 1. Your DSP-93

From: bstrick@plano.net (Bob Stricklin)  
Date: Sat, 1 Jul 1995 16:41:12 -0500  
To: hfsig@tapr.org  
Message-ID: <9507012141.AA12115@dns>  
Rendered body type: text/plain

I did mail your DSP-93 on Friday morning. It was insured so if it does not show up in a while let me know.

Also I put in a set of new docs including an operation manual. Also left in the older assembly manual in case it has something that will help you out. Can not recal if the schematics were included. If not let me know and I will mail you a set.

Bob Stricklin, N5BRG

---

## 2. Re: [HFSIG:440] Your DSP-93

From: bstrick@plano.net (Bob Stricklin)  
Date: Sun, 2 Jul 1995 08:28:13 -0500  
To: hfsig@tapr.org  
Message-ID: <9507021328.AA00123@dns>  
Rendered body type: text/plain

Sorry, I posted this note to the wrong address.

>I did mail your DSP-93 on Friday morning. It was insured so if it does not  
>show up in a while let me know.

>

>Also I put in a set of new docs including an operation manual. Also left in  
>the older assembly manual in case it has something that will help you out.  
>Can not recal if the schematics were included. If not let me know and I will  
>mail you a set.

>

>Bob Stricklin, N5BRG

>

>

Bob Stricklin, N5BRG

---

## 3. 1995 ARRL DCC (Papers Deadline)

From: Greg Jones <gjones@tenet.edu>  
Date: Mon, 3 Jul 1995 03:48:03 -0500 (CDT)  
To: aprssig@tapr.org (BBS SIG mailing), bbssig@tapr.org (BBS SIG mailing),  
hfsig@tapr.org (HF SIG mailing), netsig@tapr.org (NETSIG mailing),  
amsat-bb@amsat.org (AMSAT BB Mail Group), tapr-tnc@tapr.org  
Message-ID: <199507030848.DAA31049@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

1995 ARRL Digital Communications Conference  
September 8-10th, 1995  
Arlington, Texas (minutes from DFW airport)

\*\*\*\*\* PAPERS DEADLINE July 21st, 1995 \*\*\*\*\*

Anyone interested in digital communications is invited to submit a paper for publication in the Conference Proceedings. Presentation at the Conference is not required for publication. If you know of someone who is doing great things with digital communications, be sure to personally tell them about this! Papers are due by July 21, 1995, and should be submitted to Maty Weinberg, ARRL, 225 Main Street, Newington, CT 06111 or via the Internet to [mweinberg@arrrl.org](mailto:mweinberg@arrrl.org). Please contact Maty for detailed format requirements.

Full text is available on the Internet:  
FTP: [ftp.tapr.org](ftp://ftp.tapr.org) WWW: <http://www.tapr.org/tapr> E-Mail: [tapr@tapr.org](mailto:tapr@tapr.org)  
-----

The ARRL Digital Communications Conference is an international forum for radio amateurs in digital communications, networking, and related technologies to meet, publish their work, and present new ideas and techniques for discussion. Presenters and attendees will have the opportunity to exchange ideas and learn about recent hardware and software advances, theories, experimental results, and practical applications. The Digital Communications Conference is not just for the digital elite, but for digitally-orientated amateurs of all levels of experience.

The 1995 Digital Communications Conference will be held on September 8-10th, 1995 in Arlington, Texas. Arlington is just minutes away from the Dallas/Ft. Worth International Airport, which is located in the middle of the country, and airfares have never been cheaper for this conveniently-located conference.

In addition to the presentation of papers on Saturday, two workshops will be held during the conference. On Friday, Keith Sproul, WU2Z, will hold a workshop on APRS packet-location software. Keith is the Chair of the TAPR APRS Special Interest Group, developer of the Macintosh version of APRS, and a leader in the area of APRS technology. This is a unique opportunity to gain insight into this fast growing new digital aspect of amateur operations that combines computers, packet radio, and GPS (Global Positioning Satellites). On Sunday, Dewayne Hendricks, WD8DZP, will conduct a workshop focusing on a survey of Personal Communications Systems and their applications and use in the Amateur Radio Service. Dewayne is an expert in the area of commercial wireless systems; his company WarpSpeed Imagineering, focuses on wireless Internet. This workshop presents an opportunity to learn how Personal Communications Technology (handheld and small business wireless systems) can be used in the amateur service.

Full information on the conference and hotel information can be obtained by contacting Tucson Amateur Packet Radio, 8987-309 E. Tanque Verde Road, Tucson, AZ 85749-9399. Phone: (817) 383-0000. Fax: (817) 566-2544. Internet: TAPR@TAPR.ORG <http://www.tapr.org/tapr>

Sincerely,

Greg Jones, WD5IVD - President, Tucson Amateur Packet Radio  
Dave Wolf, W05H - President, Texas Packet Radio Society

---

#### 4. Re: [HFSIG:436] Re: HFSIG activities

From: JAL0CHA@chopin.ifj.edu.pl  
Date: Tue, 4 Jul 1995 13:42 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <9C6FD357C0222014@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>It may be useful to use a Hilbert transform especially if your demodulator  
>require I/Q paths, i.e. doing complex FFT's. You do need, however, some good  
>low pass filters to go with it. All these computational blocks eats at your  
>machine ticks, however. If you can fit it in, that will be excellent. We had  
>some discussion about this topic under "HF channel simulator".

So I am following this path right now... I did a major redesign so now the receiver stages are:

1. Sampling at 9600 Hz
2. Bandpass filter 400-2900 Hz (1650+/-1250 Hz),  
sin(x)/x and cos(x)/x parts, complex output  
- is this called the Hilbert transform ?
3. Downsampling by 3 => makes the sampling rate and bandwidth of 9600/3=3200 Hz  
(I can reduce the sampling rate down to the bandwidth because I analyze a complex signal)
4. Complex numeric oscillator and mixer.
5. Sliding window + complex FFT (64 points)  
32 points between symbols => 3200/32 = 100 symbols/sec.  
FFT bins are 50 Hz wide, intended carrier spacing is 150 Hz.

Such design let me do some savings on the FFT CPU cycles and storage requirements so the current code is more compact.

The transmitter stages are very much similar (they go backwards)  
only the oscillator+mixer part is avoided.  
Provision for symbol timing adjustment at the receiver is already there.

The "band plan": 15 data carriers spaced by 150 Hz.  
First carrier at 600 Hz, last at 2700 Hz. Each carrier is DQPSK modulated as 100 baud (2 bits/symbol). Total data rate is  $15 \times 100 \times 2 = 3000$  bps.  
Or shall I go to 30 carriers at 50 baud spaced by 75 Hz ?

I plan to make the modem send some sort of preamble which would enable fast tuning and symbol-sync. acquisition. For example send every second carrier unmodulated for tuning preamble, then modulate the carriers by inverting the phase every symbol for fast symbol-sync.

The current code (DSPCARD4/EVM56002) contains all the stages I described and it seems to work ! Yesterday I made the modem send the preambles I described. My expectation is that the modem could auto-tune within  $\pm 50$  Hz without any loss in performance - what do people think, is that enough ?

>For interest sake, could you perhaps tell us a bit more about the FREQ4  
>software - where to obtain it etc. These analytical methods and tools are  
>useful to know about.

A good revision of FFT software was given already on this list.  
I can only confirm that FREQ4 looks very nice, unfortunately  
I don't have an SVGA to run the real nice version :-)  
However one there is one thing I miss in the FREQ4: the spectrum averaging feature. There is a "peak" and "decay" display mode but that's not really averaging.

Pawel

---

## 5. Re: [HFSIG:443] Re: HFSIG activities

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Tue, 04 Jul 1995 17:39:39 -0800  
To: hfsig@tapr.org  
Message-ID: <9507050128.AA13889@ucs.orst.edu>  
Rendered body type: text/plain

Pawel,

>So I am following this path right now... I did a major redesign so now  
>the receiver stages are:  
>  
>1. Sampling at 9600 Hz  
>2. Bandpass filter 400-2900 Hz ( $1650 \pm 1250$  Hz),  
> sin(x)/x and cos(x)/x parts, complex output  
> - is this called the Hilbert transform ?

Yes, same filters, just 90 degrees phase shifted.

>3. Downsampling by 3 => makes the sampling rate and bandwidth of  $9600/3 = 3200$  Hz  
> (I can reduce the sampling rate down to the bandwidth because I analyze  
> a complex signal)

Good idea - you are picking up some processing gain doing this.

>4. Complex numeric oscillator and mixer.

I suppose this is to go down to zero IF.

>5. Sliding window + complex FFT (64 points)  
> 32 points between symbols =>  $3200/32 = 100$  symbols/sec.  
> FFT bins are 50 Hz wide, intended carrier spacing is 150 Hz.

Sounds quite conservative.

Your demodulator is beginning to sound much like Adrian's one for Piccolo.  
This very much along the lines I have seen described for several commercial designs. You are certainly on the right track!

>

>Such design let me do some savings on the FFT CPU cycles  
>and storage requirements so the current code is more compact.  
>

I'm not sure why this is the result. Could you tell use why?

>The transmitter stages are very much similar (they go backwards)  
>only the oscillator+mixer part is avoided.



>Provision for symbol timing adjustment at the receiver is already there.  
 >  
 >The "band plan": 15 data carriers spaced by 150 Hz.  
 >First carrier at 600 Hz, last at 2700 Hz. Each carrier is QPSK modulated  
 >as 100 baud (2 bits/symbol). Total data rate is 15\*100\*2=3000 bps.  
 >Or shall I go to 30 carriers at 50 baud spaced by 75 Hz ?

This is well worth a try - 100 baud is approaching the upper limit of what is feasible due to multipath. If this does not work out as you like, go for the the 30 carriers at 50 baud.

>  
 >I plan to make the modem send some sort of preamble which would enable  
 >fast tuning and symbol-sync. acquisition. For example send every second  
 >carrier unmodulated for tuning preamble, then modulate the carriers  
 >by inverting the phase every symbol for fast symbol-sync.  
 >

This is an extensive topic - we can talk about it a lot. Your suggestion sounds like a good one - if you can make it work. For such a synchronization preamble, some folks use say two tones from your set, choosen as far apart as possible, and they are sent for a duration of say 5 baud times. One tone is unmodulated and used for doppler estimation and correction. The other tone is phase shifted by pi at each baud interval and thus gives you symbol synch. It is also common that doppler tone is sent with say double the magnitude (6-7dB) than the symbol sync. tone or any of the other carriers for that matter. There should be no confusion about it.

Symbol synch/Doppler estimation may be done several ways - a clever idea is to use you sliding FFT demodulator with a pair of tones that alternates on/off at the symbol rate. Then estimate your amount of frequency/timing error using the following maximum-likelihood estimator for a particular timing interval:

$$\text{Error} = (r2-r1)/(2(r1+r2))$$

Where r1 and r2 are the Fourier magnitudes of the two alternating tones TIMED AT 1/2 the timing interval. For perfect sych. Error=0, and Error=+/-0.5 at an offset of 1/2T. The explantion is a little involved to expand on but is very elegantly presented in:

The design of a low data rate MFSK communication system. By Henry D. Chadwich and James C. Springett. IEEE Trans. on Communications Tech. VOL: COM-18, No.6 (December 1970) pp.740-750.

>The current code (DSPCARD4/EVM56002) contains all the stages I described  
 >and it seems to work ! Yesterday I made the modem send the preambles  
 >I described. My expectation is that the modem could auto-tune within +/- 50Hz  
 >without any loss in performance - what do people think, is that enough ?

That is remarkable. Let us know when you are ready with an interim version for testing!

>A good revision of FFT software was given already on this list.  
 >I can only confirm that FREQ4 looks very nice, unfortunately  
 >I don't have an SVGA to run the real nice version :-)

What are you missing? The display card or the monitor? I have a card that you are welcome to have.

>However one there is one thing I miss in the FREQ4: the spectrum  
 >averaging feature. There is a "peak" and "decay" display mode but  
 >that's not really averaging.  
 >

Keep up the good work. Wish I could find more time to join in experiments - in a little while I hope thigs will ease up a bit.

73's

--Johan, KC7WW

## 6. Re: [HFSIG:444] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl  
Date: Wed, 5 Jul 1995 12:07 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <584D5BEC4022A761@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>>2. Bandpass filter 400-2900 Hz (1650+/-1250 Hz),  
>> sin(x)/x and cos(x)/x parts, complex output  
>> - is this called the Hilbert transform ?  
>  
>Yes, same filters, just 90 degrees phase shifted.

A question here: how do I design a good anti-alias filter ?  
good = not too many taps and sufficient alias band rejection.  
For the moment I calculate it with  $\sin(x)/x * \sin(x)^2$  or another window.  
I guess I should get some FIR filters design software.

>>4. Complex numeric oscillator and mixer.  
>

>I suppose this is to go down to zero IF.

Not really (if I understand the topic right) - normally the NCO  
runs at frequency=0 (thus it doesn't affect the signal).  
When I need to tune say up 10 Hz I set the NCO to 10 Hz  
and when I want to tune down 10 Hz I set it to -10 Hz.

For zero-IF you set the NCO to the center of your passband (1650 Hz) ?

>>Such design let me do some savings on the FFT CPU cycles  
>>and storage requirements so the current code is more compact.  
>>  
>  
>I'm not sure why this is the result. Could you tell use why?

The previous design: for 12 tones I had to do a 128 point `_real_` FFT.  
I don't have a code for real FFT and I only have one for complex FFT  
(the one taken from the Motorola bulletin board). Thus I was doing twice  
the work and after that half of the data was to be put away.  
With the current design I do a 64 point `_complex_` FFT for 15 tones  
and all the resulting bins contain usefull information.  
There are some other savings done as well, for example I only store  
the bins I want; before I was storing the whole band for several  
samples "just in case" I need it for re-tuning - many arrays got smaller,  
only the CODEC's buffer stayed huge :-)

>>The "band plan": 15 data carriers spaced by 150 Hz.  
>>First carrier at 600 Hz, last at 2700 Hz. Each carrier is DQPSK modulated  
>>as 100 baud (2 bits/symbol). Total data rate is  $15*100*2=3000$  bps.  
>>Or shall I go to 30 carriers at 50 baud spaced by 75 Hz ?  
>  
>This is well worth a try - 100 baud is approaching the upper limit of what  
>is feasible due to multipath. If this does not work out as you like, go for  
>the the 30 carriers at 50 baud.

This is not a problem for the code, only the mis-tune tolerance  
gets decreased by 2 (+/-25Hz ?) unless you do some clever tricks :-)

>>I plan to make the modem send some sort of preamble which would enable  
>>fast tuning and symbol-sync. acquisition. For example send every second  
>>carrier unmodulated for tuning preamble, then modulate the carriers  
>>by inverting the phase every symbol for fast symbol-sync.  
>  
>This is an extensive topic - we can talk about it a lot. Your suggestion  
>sounds like a good one - if you can make it work. For such a synchronization  
>preamble, some folks use say two tones from your set, choosen as far apart  
>as possible, and they are sent for a duration of say 5 baud times. One tone  
>is unmodulated and used for doppler estimation and correction. The other  
>tone is phase shifted by pi at each baud interval and thus gives you symbol  
>synch. It is also common that doppler tone is sent with say double the  
>magnitude (6-7dB) than the symbol sync. tone or any of the other carriers  
>for that matter. There should be no confusion about it.

I like more the idea of many tones. I want to keep the modem resistant  
to CW-type jamming. If I use one tone, and the jamming falls onto it...

By the way I wonder whether I should keep symbol sync. frozen once  
acquired in the preamble or shall I still follow it during the data phase ?

If I run my sampling on crystals this seems to be not really needed...  
or there are some ionospheric effects which could "magically" shift the timing ?  
Same thing about tuning correction...  
Anyway, \_not\_ following the symbol timing would do major code saving:  
I don't need to sample at twice the symbol rate.  
Following the mis-tune doesn't cost much: you only sum up the phase error  
and if you see a constant trend you correct the NCO.

```
>Symbol synch/Doppler estimation may be done several ways - a clever idea is
>to use you sliding FFT demodulator with a pair of tones that alternates
>on/off at the symbol rate. Then estimate your amount of frequency/timing
>error using the following maximum-likelihood estimator for a particular
>timing interval:
>
> Error = (r2-r1)/(2(r1+r2))
```

I don't understand yet, but I will think about it :-)

My plan was to synchronize on the alternating phase preamble...  
However it would be just simpler to have one common preamble  
to do tune and sync. at same time. The one you suggest may be  
a good candidate.

```
>>A good revision of FFT software was given already on this list.
>>I can only confirm that FREQ4 looks very nice, unfortunately
>>I don't have an SVGA to run the real nice version :-)
>
>What are you missing? The display card or the monitor? I have a card that
>you are welcome to have.
```

My whole PC needs a major upgrade, but I'm going to get a new one  
sooner or later, don't worry :-)

Pawel

---

## 7. Re: [HFSIG:445] Re: HFSIG activities

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>  
Date: Wed, 05 Jul 95 09:44:47 nst  
To: hfsig@tapr.org  
Message-ID: <9506058049.AA804962719@ccsmtp.udc.neweast.ca>  
Rendered body type: text/plain

Pawel,

```
>>>The "band plan": 15 data carriers spaced by 150 Hz.
>>First carrier at 600 Hz, last at 2700 Hz. Each carrier is DQPSK modulated
>>>as 100 baud (2 bits/symbol). Total data rate is 15*100*2=3000 bps.
>>>Or shall I go to 30 carriers at 50 baud spaced by 75 Hz
>? >
>>This is well worth a try - 100 baud is approaching the upper limit of what
>>is feasible due to multipath. If this does not work out as you like, go for
>>the the 30 carriers at 50 baud.
>
>This is not a problem for the code, only the mis-tune tolerance
>gets decreased by 2 (+/-25Hz ?) unless you do some clever tricks :-)
```

If you use less tones for sync, then you can regain the mis-tune  
tolerance. If the receiver knows which tones to expect, and say  
that you are using 4 of the possible 15 spaced at 200Hz (300Hz?),  
then you get 4\* the spacing tolerance: 100Hz (150Hz). If my math  
is reasonable a good balance between jamming resistance and  
mis-tune tolerance can be had by using 1/2, 1/4, or 1/8 or the  
tones during tuning.

If you use symbol shaping (raised cosine) and alternate the phase  
each symbol, a rectifier feeding a 100Hz Narrow 2nd order (3Hz  
wide) BPF, then a DPLL, will give quick lockon.

```
>>>I plan to make the modem send some sort of preamble which would enable
>>>fast tuning and symbol-sync. acquisition. For example send every second
>>>carrier unmodulated for tuning preamble, then modulate the carriers
>>>by inverting the phase every symbol for fast symbol-sync.
>>
>>This is an extensive topic - we can talk about it a lot. Your suggestion
>>sounds like a good one - if you can make it work. For such a synchronization
>>preamble, some folks use say two tones from your set, choosen as far apart
>>as possible, and they are sent for a duration of say 5 baud times. One tone
>>is unmodulated and used for doppler estimation and correction. The other
>>tone is phase shifted by pi at each baud interval and thus gives you symbol
```

>>synch. It is also common that doppler tone is sent with say double the  
>>magnitude (6-7dB) than the symbol sync. tone or any of the other carriers  
>>for that matter. There should be no confusion about it.

>I like more the idea of many tones. I want to keep the modem resistant  
>to CW-type jamming. If I use one tone, and the jamming falls onto  
>it...

>By the way I wonder whether I should keep symbol sync. frozen once  
>acquired in the preamble or shall I still follow it during the data phase ?  
>If I run my sampling on crystals this seems to be not really needed...  
>or there are some ionospheric effects which could "magically" shift the timing  
>? Same thing about tuning correction...  
>Anyway, not following the symbol timing would do major code  
>saving: I don't need to sample at twice the symbol rate.  
>Following the mis-tune doesn't cost much: you only sum up the phase error  
>and if you see a constant trend you correct the NCO.

Careful of freezing the symbol sync. Crystals are NOT as reliable as you think, especially in places varied by temperature (you in death valley and your buddy in the Artic <g>). 100ppm (100 parts per million is a common error), so 100symbols/sec at 100/1,000,000 = 10,000 symbols before you are missaligned by 1 bit (or 5000symbols if 1 end is 100ppm high and the other 100ppm low). I suspect the modem design could not tolerate more than 2 to 4 FFT bins of missalignment on a symbol, so 5000 \* 1/64 = 78symbols worst case, 10000\*4/64 = 625 symbols best case.  
So, unless your packets are short, you should run continuous clock alignment. One common method is to use wideband and narrow band alignment. i.e. allow wide clock adjustment during sync up (8/64 per symbol), and only a small alignment during packet reception (1/64 per symbol, or even fractional adjusts of say (1/4)/64 (-4 in a row to high/low before correct by 1 clock tick).

Tuning tolerance is probably similiar. I suspect that unless you are in a jet doing fast turns towards and away from the source, you will have very minor doppler (due to ionosphere layer movements, or during the switch from one path to another path in multipath). If your CPU can handle it, set it to track at probably between 0.1Hz and 1 Hz/symbol (arbitrary values, sorry no test data).

So if you can course tune on a limited tone preamble: adjust for >50Hz error to < 10Hz error (Adjust limited to 1/2 the spacing between tones present). Then fine tune during the end of the preamble and during the data (adjust limited by 1/2 the tone spacing). This would indeed be a nice system. I suspect my company (Ultimateast, 709-576-4747, ask for Dwight Howell or John Mackey, say Paul Russell referred you) would definitely consider buying an implementation (I'm moving to a new job and won't be here to do it myself).

Paul Russell  
paulr@neweast.ca (for this and next week)

---

## 8. Re: [HFSIG:446] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl  
Date: Wed, 5 Jul 1995 15:44 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <76B89AC62022A761@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>If you use less tones for sync, then you can regain the mis-tune  
>tolerance.

Yes, I had this in mind...

>If the receiver knows which tones to expect, and say  
>that you are using 4 of the possible 15 spaced at 200Hz (300Hz?),  
>then you get 4\* the spacing tolerance: 100Hz (150Hz). If my math  
>is reasonable a good balance between jamming resistance and  
>mis-tune tolerance can be had by using 1/2, 1/4, or 1/8 or the  
>tones during tuning.

1/4 is better than 1/2 and 1/8 for one "trivial" reason:  
if you want to keep same RMS while transmitting 1/4th of the tones  
you should increase the tone's amplitudes by a factor of 2.  
For 1/2 or 1/8 this factor becomes a bit more ugly :-)

For the moment I put in 4 "tuning" tones: 750, 1350, 1950, 2550 Hz  
(600 Hz spacing). Thus in principle I could have +/-300Hz tuning tolerance.  
But... my passband is not that wide. I would need to make more "spare"  
bandwidth in the transceiver audio. This is another issue to discuss:

I am trying to fit it 2500 Hz of usefull spectrum. Isn't it too much ?  
We could as well use only 2000 Hz with 16 tones (like one military standard does) and leave 200-300 Hz on both sides for tuning corrections and equipment imperfections ?

>If you use symbol shaping (raised cosine) and alternate the phase  
>each symbol, a rectifier feeding a 100Hz Narrow 2nd order (3Hz  
>wide) BPF, then a DPLL, will give quick lockon.

You mean avoiding the FFT and putting the time-domain samples straight into the rectifier. This might not be as good as looking at selected tones after the FFT (jamming resistance again) but it's a good idea.

>So, unless your packets are short, you should run continuous clock  
>alignment. One common method is to use wideband and narrow band alignment.  
>i.e. allow wide clock adjustment during sync up (8/64 per symbol), and only a  
>small alignment during packet reception (1/64 per symbol, or even fractional  
>adjusts of say (1/4)/64 (-4 in a row to high/low before correct by 1 clock  
>tick).

OK. I will run a slow clock adjustment during the data phase.  
I was taking a very optimistic 10E-5 crystal stability for my calculations :-)  
but if they can be 10E-4...

>Tuning tolerance is probably similiar. I suspect that unless you are in a jet  
>doing fast turns towards and away from the source, you will have very minor  
>doppler (due to ionosphere layer movements, or during the switch from one  
>path to another path in multipath). If your CPU can handle it, set it to  
>track at probably between 0.1Hz and 1 Hz/symbol (arbitrary values, sorry no  
>test data).

Infact for the Doppler shift I was afraid more of transceiver's  
instabilities... but I should do a slow adjustments anyway.  
General conclusion: do both adjustments all the time !

>So if you can course tune on a limited tone preamble: adjust for >50Hz error  
>to < 10Hz error (Adjust limited to 1/2 the spacing between tones present).  
>Then fine tune during the end of the preamble and during the data (adjust  
>limited by 1/2 the tone spacing). This would indeed be a nice system.

I had in mind coarse tuning adjustment during the tuning preamble  
and when the symbol sync. preamble comes, I could refine the tuning  
while doing symbol sync. - this is not much a problem.  
I suspect this will be the major problem (at least for me) to make the code  
do all these things at the right time :-) Tuning, symbol sync., data coding  
I know how to do in principle...

>I suspect my company (Ultimateast, 709-576-4747, ask for Dwight Howell or John  
>Mackey, say Paul Russell referred you) would definitely consider buying an  
>implementation (I'm moving to a new job and won't be here to do it myself).

And what happens when I sell the implementation ? Would it be still  
available to amateur radio ? :-)

Pawel

---

## 9. Re: [HFSIG:447] Re: HFSIG activities

From: "Paul Russell" <paulr@hagar.udc.neweast.ca>  
Date: Thu, 06 Jul 95 11:06:04 nst  
To: hfsig@tapr.org  
Message-ID: <9506068050.AA805053979@ccsmtp.udc.neweast.ca>  
Rendered body type: text/plain

>1/4 is better than 1/2 and 1/8 for one "trivial" reason:  
>if you want to keep same RMS while transmitting 1/4th of the tones  
>you should increase the tone's amplitudes by a factor of 2.  
>For 1/2 or 1/8 this factor becomes a bit more ugly :-)

Note: The individual phase sets for each tone don't have to use  
the same reference in DPSK.

so Tone 1 could use 10, 100, 190, 280      QDPSK  
    tone 2 could use 350, 80, 170, 260

etc.

An optimization program (weeks of background processing time) can  
be written (did something similar for a 4 channel FSK) that will  
try to optimize a good reference for each tone phase set. You  
have to calculate the worst superposition of the output symbol  
for all possible tone phase combinations, while varying each  
starting tone phase reference by 1 sample in the tone generator.

When done you can gain upto a factor of 2 on a peak to average signal amplitude (4\* power).

As long as each set of phases for each tone is ok (DBPSK @180, DQPSK @90, 8 @ 45...) the sets can have independent offsets, and there are combinations of offsets for a tone set that will minimize the peak for the whole symbol set by a factor of about 2:1.

>For the moment I put in 4 "tuning" tones: 750, 1350, 1950, 2550 Hz  
>(600 Hz spacing). Thus is principle I could have +/-300Hz tuning tolerance.  
>But... my passband is not that wide. I would need to make more "spare"  
>bandwidth in the transceiver audio. This is another issue to discuss:  
>I am trying to fit it 2500 Hz of useful spectrum. Isn't it too much ?  
>We could as well use only 2000 Hz with 16 tones (like one military standard  
>does) and leave 200-300 Hz on both sides for tuning corrections  
>and equipment imperfections ?

I would definitely recommend going to a narrower minimum requirement, not all Hams may have radios with nice passbands and the more general the design the more potential users. Best radio I've seen (300-2750Hz) worst (450-2550), so in my experience if you only use about (550-2550) it might be best. Or if a protocol can be configured to test and drop/ignore the outside tones in some connections (like a Trailblaser telephone modem does). Tones dropped figured during link setup or by pre-programming? Maybe the initial call done using a subset (1000-2000Hz only) then after contact eval exactly how wide is available? I know this is costly in development time, but it may pay off later in easier use.

>>If you use symbol shaping (raised cosine) and alternate the phase  
>>each symbol, a rectifier feeding a 100Hz Narrow 2nd order (3Hz  
>>wide) BPF, then a DPLL, will give quick lockon.  
>  
>You mean avoiding the FFT and putting the time-domain samples straight  
>into the rectifier. This might not be as good as looking at selected  
>tones after the FFT (jamming resistance again) but it's a good idea.

Constant tone jamming or slow fading falls out as you are only tracking the average AC component of the signal power at the BPF. Jamming with a pulsing signal at near the baud rate would still be a problem, but random data transmitted over the signal in similiar modulation will confuse almost everything except maybe spread spectrum.

>And what happens when I sell the implementation ? Would it be still  
>available to amateur radio ? :-)

Depends on how you wrote the contract (so yes). This company puts most of the propriatary stuff in the protocols, channel evaluation, and interconnection to other systems, instead of the modulation anyway. Just a thought, can't say at this point whether it would be worth the effort. But if you do have something in say 6 months give them a call...

Pawel

---

## 10. mail\_archive

From: Greg Jones <gjones@tenet.edu>  
Date: Sat, 8 Jul 1995 09:38:26 -0500 (CDT)  
To: bbssig@tapr.org (BBS SIG mailing), netsig@tapr.org (NETSIG mailing),  
hfsig@tapr.org (HF SIG mailing), aprssig@tapr.org (BBS SIG mailing)  
Message-ID: <199507081438.JAA23566@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

The mail archives for all the SIGs are now updated nightly.

They can be gotten by ftp in each SIG area under mail\_archive

ftp://ftp.tapr.org/tapr/SIG/

In addition, the mail\_archive files have been available from the TAPR listserv.

There are links to the SIGs in the TAPR web page.  
<http://www.tapr.org/tapr>

Cheers - Greg

---

## 11. ANDVT TACTERM Documentation

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Mon, 10 Jul 1995 16:52:19 +0200 (MET DST)  
To: HFSIG TAPR <hfsig@tapr.org>  
Message-ID: <Pine.ULT.3.91.950710154436.14265A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

Hi all,

To keep the subject on multitone modems hot and as an aid in cracking open the outstanding problems on Pawel's multitone design like synchronization and error correction coding I have the following minor contributions to make.

I have been reading up on what the commercial and military users have done and are doing in the multitone field and I just uploaded a ZIP archive on TAPR's FTP server (ftp.tapr.org) in the HFSIG upload directory: /tapr/SIG/hfsig/upload with some information on the protocols used in the ANDVT TACTERM AND AN/USQ-83(V) modems.

The archive is called andvt.zip and here is the contents of the andvt.txt file:

This archive contains a total of three files.

- readme            Which you are reading now
- andvt.asc        ASCII version of The Variable Speed HF Modem
- andvt.doc        WORD 5.5 version of The Variable Speed HF Modem

If any of the two files andvt.asc and andvt.doc are distributed in a different format than the present ZIP archive, please make sure to include this readme file.

The documentation of The Variable Speed HF Modem (VSM) has been extracted almost verbatim from David L. Tate's report "The Variable Speed HF Modem, Modulator Design". NRL Report 9169, March 20, 1989; NTIS Report No.: AD-A244 891. The parts extracted concerns mainly the description of the protocols used by the ANDVT TACTERM AND AN/USQ-83(V) modems, hence the file names andvt.asc and andvt.doc.

This writeup on the VSM has been made by Frode Weierud, LA2RL, with the authorisation of the original author David Tate. On the question of copyright etc. David Tate have instructed me as follows:

Dear Mr. Weierud,

The report titled "The Variable Speed HF Modem Modulator Design" is not copyrighted and you are free to distribute any portion of it as you wish. If you include any of it in any published report, I would ask that you reference it as a professional courtesy.

- David Tate

Therefore the two documents andvt.asc and andvt.doc are also free of any copyrights, but I insist that the original author's request for proper referencing is adhered to also in the case of information taken from my two documents andvt.asc and andvt.doc.

Frode Weierud, LA2RL  
E-mail : frode@dxcern.cern.ch

-----

I hope some of you might get some good ideas about how to handle the symbol and bit synchronization in our multitone modem when seeing how it is done in the ANDVT TACTERM modem.

Another issue that surely will come up later is error correction coding. I have recently re-read a paper that deals with this and which I find is well written and gives a lot of useful information on error correction codes used with parallel and serial tone modems on HF. The reference is:

P.G. Farrell and R.M.F Goodman, "Soft-decision error control for h.f. data transmission", IEE Proc., Vol. 127, Part F, No.5, Oct. 1980. (NB! This is the English Institute of Electrical Engineers)

73's Frode, LA2RL

Frode Weierud                      Phone : 41 22 7674794  
CERN, SL                              Fax : 41 22 7679185  
CH-1211 Geneva 23, Switzerland   E-mail : frode@dxcern.cern.ch

---

## 12. Re: [HFSIG:450] ANDVT TACTERM Documentation

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Mon, 10 Jul 95 21:11:51 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sVUng-0002EZC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

In Frode's message of 10 Jul 1995 at 1007 CDT, he writes:

> To keep the subject on multitone modems hot and as an aid in cracking  
> open the outstanding problems on Pawel's multitone design like  
> synchronization and error correction coding I have the following minor  
> contributions to make.  
>  
> I have been reading up on what the commercial and military users have  
> done and are doing in the multitone field and I just uploaded a ZIP  
> archive on TAPR's FTP server (ftp.tapr.org) in the HFSIG upload  
> directory: /tapr/SIG/hfsig/upload with some information on the protocols  
> used in the ANDVT TACTERM AND AN/USQ-83(V) modems.

The ANDVT TACTERM is of rather old design and I believe was the first implementation of the 39 parallel tone modem of MIL-STD-188-110A. If my memory serves me correctly, the TACTERM is over 10 years old.

For any who read about this device, on voice operation the modem is the MIL-STD 39 tone modem with all the FEC, etc features of the MIL-STD turned on. On data, the modem uses the MIL-STD 16 tone modem format which does \*not\* have FEC. Both modems run at 2400 BPS on HF.

I used the TACTERM on HF data at 2400 BPS (and voice) before and during Desert Shield/Storm and it beats the pants off of 110 baud (and 300 baud) ASCII, packet, AMTOR and the like...and that is without FEC. I can just imagine what the 16 tone modem would do with a little FEC.

This would be a good model to duplicate using DSP and a soundcard as a first effort in getting high speed data on HF.

Walt

---

## 13. Re: [HFSIG:451] Re: ANDVT TACTERM Documentation

From: JALOCHA@chopin.ifj.edu.pl  
Date: Wed, 12 Jul 1995 11:58 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <D73B85306022A758@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>I used the TACTERM on HF data at 2400 BPS (and voice) before and  
>during Desert Shield/Storm and it beats the pants off of 110 baud (and  
>300 baud) ASCII, packet, AMTOR and the like...and that is without FEC.

You mean TACTERM beats other protocols in the sense of data rate or better performance in weak propagation conditions ?

>This would be a good model to duplicate using DSP and a soundcard as  
>a first effort in getting high speed data on HF.

I am on a very close track actually, if I change decimation ratio in my design from 3 to 4 I get exactly the tone spacing and symbol rate used by TACTERM.

One interesting detail of TACTERM I noticed: the tune-in preamble uses tones from outside the data band... and I don't see a good reason for this. Why send two tones where you are not going to send any data ? I would rather filter out this part of audio.

Progress report on my code - as it is now it does the following:

Tx:

- sends four tuning tones for 32 symbols (0.32 second).
- modulates the tuning tones by alternating the phase for 32 symbols. This is for the receiver to get symbol-sync.
- switches to 15 tones and sends 100 "random" symbols.
- sends 16 symbols of jamming data so the receiver's DCD switches off



quickly.

Rx:

- Listens and waits for the tuning tones.  
When found and present for some minimal time the Rx finds out the freq. error and switches to sync. wait mode.
- Waits until the tones become bi-modulated in phase.  
If they don't within some timeout the Rx goes back to "listen for tune" mode.  
If they do, and stay modulated for some minimal time, the Rx finds out the symbol timing, freq. error and switched to data mode. This part is not yet all done and is right now under developement.
- Next, the Rx should receive the data, and do little freq. and symbol-sync. adjustments while monitoring the DCD condition. When DCD drops, the Rx switches to "listen for tune".

Pawel

---

## 14. Re: [HFSIG:451] Re: ANDVT TACTERM Documentation

From: claude@bauv106.bauv.unibw-muenchen.de (Claude Frantz)  
Date: Wed, 12 Jul 95 13:02:18 MET DST  
To: hfsig@tapr.org  
Message-ID: <9507121102.AA00417@bauv106.bauv.unibw-muenchen.de>  
Rendered body type: text/plain

According to WALT DUBOSE - K5YFW:

- > I used the TACTERM on HF data at 2400 BPS (and voice) before and
- > during Desert Shield/Storm and it beats the pants off of 110 baud (and
- > 300 baud) ASCII, packet, AMTOR and the like...and that is without FEC.
- > I can just imagine what the 16 tone modem would do with a little FEC.
  
- > This would be a good model to duplicate using DSP and a soundcard as
- > a first effort in getting high speed data on HF.

Have you any information about the technology used by this TACTERM (hardware, software) ?

Claude

---

## 15. Re: [HFSIG:452] Re: ANDVT TACTERM Documentation

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Wed, 12 Jul 1995 14:24:38 +0200 (MET DST)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.950712140350.9690B-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Wed, 12 Jul 1995 JALOCHA@chopin.ifj.edu.pl wrote:

- >
- > I am on a very close track actually, if I change decimation ratio
- > in my design from 3 to 4 I get exactly the tone spacing and symbol rate
- > used by TACTERM.
- >
- > One interesting detail of TACTERM I noticed: the tune-in preamble
- > uses tones from outside the data band... and I don't see a good
- > reason for this. Why send two tones where you are not going to send
- > any data ? I would rather filter out this part of audio.
- >

I wonder if it could be implementation related. The HF radio will have a 300 - 3000Hz IF bandwidth so the four Doppler tones will fit OK in this bandwidth. The two middle tones used for tuning (Doppler tones) correspond to data tone No. 6 and No.12 (1462.5 Hz and 2137.5 Hz). The distance between the two tones is 675 Hz, which is the distance between all the four Doppler tones. 675 Hz is also the distance between the three frame sync tones, which correspond to data tones No. 3, 9 and 15. This distance of 675 Hz is 6 times the data tone distance which is 112.5 Hz.

I therefore suppose that they wanted to have a different set of tones for Doppler preamble and frame preamble, but that they wanted to keep the same bin distance of 675 Hz. The symmetrical selection of these tones with respect to the data tones then means two of the Doppler tones will fall slightly outside the boundary of the data tone set.

Pawel, congratulation with the tremendous progress on your parallel tone modem. I am curious to see what will happen when you modulate a SSB rig with this waveform. As Paul, I am somewhat afraid we will run into problems with the IF bandwidth of most HAM rigs. I just read a review of the ICOM IC-736 HF rig yesterday, where they proudly announce very steep

IF filters with 2.1 kHz bandwidth, exactly what we don't want, :-)

I know this is not a real problem, but I am afraid we probably will have to reduce the number of tones to be sure to fit nicely within a 2.1 kHz IF bandwidth.

73's Frode

Frode Weierud                      Phone : 41 22 7674794  
CERN, SL                              Fax : 41 22 7679185  
CH-1211 Geneva 23, Switzerland   E-mail : frode@dxcern.cern.ch

---

## 16. Re: [HFSIG:454] Re: ANDVT TACTERM Documentation

From: JALOCHA@chopin.ifj.edu.pl  
Date: Wed, 12 Jul 1995 15:13 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <F2734C1AC022A758@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>I am curious to see what will happen when you modulate a SSB rig  
>with this waveform. As Paul, I am somewhat afraid we will run into  
>problems with the IF bandwidth of most HAM rigs. I just read a review of  
>the ICOM IC-736 HF rig yesterday, where they proudly announce very steep  
>IF filters with 2.1 kHz bandwidth, exactly what we don't want, :-)

Did they say where are the audio band edges ?  
Why don't they make a rig with 1 kHz bandwidth ? :-)  
Actually it's easier to make a narrow crystal filter than a wide one.  
But maybe the IC-738 filter's width is still switchable ?  
I know there are rigs with 1.8 kHz bandwidth... if we want to keep  
compatibility with these, we have to squeeze the tones even more.

>I know this is not a real problem, but I am afraid we probably will have  
>to reduce the number of tones to be sure to fit nicely within a 2.1 kHz  
>IF bandwidth.

I expect this to be a rather minor operation.  
I would rather stay with 16 tones but reduce the spacing to 112.5 Hz  
and the baud rate to 75 - are here we came to TACTERM :-)  
16 is a nice number for FEC-codes... like Walsh functions.  
Later we can try 32 tones with 37.5 baud each.

By the way is there an optimal symbol rate for HF and DQPSK ?  
If too high, the multi-paths come into effect,  
if too low, the phase incoherency affects the data...  
so where is the optimum ?

Pawel

---

## 17. Re: [HFSIG:452] Re: ANDVT TACTERM Documentation

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Wed, 12 Jul 95 23:49:30 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sWGDl-0002G1C@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

In Pawel's message of 12 Jul 1995 at 0501 CDT, he writes:  
> >I used the TACTERM on HF data at 2400 BPS (and voice) before and  
> >during Desert Shield/Storm and it beats the pants off of 110 baud (and  
> >300 baud) ASCII, packet, AMTOR and the like...and that is without FEC.  
>  
> You mean TACTERM beats other protocols in the sense of data rate  
> or better performance in weak propagation conditions ?

We had a 300 baud Bell 102 modem device (terminal) and  
PK-232s and other Military modems for 50/100/110/300 baud...  
running RATT, ASCII, AX.25, AMTOR.

2400 BPS was faster than the others and had acceptable  
thruput during poor signal conditions. In poor conditions we  
got near 2400 BPS thrupt vs no thrupt with the 110/300 baud  
modems/protocols. AMTOR/SITOR worked as well as the TACTERM  
in poor conditions but the thrupt of AMTOR/SITOR was no  
comparison.

I didn't try the TACTERM an any bit rate less than 2400...  
maybe 1200 once.

>

> >This would be a good model to duplicate using DSP and a soundcard as  
> >a first effort in getting high speed data on HF.  
>  
> I am on a very close track actually, if I change decimation ratio  
> in my design from 3 to 4 I get exactly the tone spacing and symbol rate  
> used by TACTERM.  
>  
> One interesting detail of TACTERM I noticed: the tune-in preamble  
> uses tones from outside the data band... and I don't see a good  
> reason for this. Why send two tones where you are not going to send  
> any data ? I would rather filter out this part of audio.

Good show. I'm sure that the 10+ years of technology since  
the TACTERM was designed/produced should all for improvements  
in the modem protocol. I do feel that any new modem protocol  
should include FEC.

If I remember correctly, 2400 BPS is the thruput rate but the  
modulation bit rate is 3100 BPS on the 39 tone...about 1/3 of  
the data is for the FEC. What would FEC do to a 16 tone  
modem?

>  
> Progress report on my code - as it is now it does the following:  
>  
> Tx:  
> - sends four tuning tones for 32 symbols (0.32 second).  
> - modulates the tuning tones by alternating the phase for 32 symbols.  
> This is for the receiver to get symbol-sync.  
> - switches to 15 tones and sends 100 "random" symbols.  
> - sends 16 symbols of jamming data so the receiver's DCD switches off  
> quickly.  
>  
> Rx:  
> - Listens and waits for the tuning tones.  
> When found and present for some minimal time the Rx finds out  
> the freq. error and switches to sync. wait mode.  
> - Waits until the tones become bi-modulated in phase.  
> If they don't within some timeout the Rx goes back to "listen for tune" mode.  
> If they do, and stay modulated for some minimal time, the Rx finds out  
> the symbol timing, freq. error and switched to data mode. This part is  
> not yet all done and is right now under developement.  
> - Next, the Rx should receive the data, and do little freq. and symbol-sync.  
> adjustments while monitoring the DCD condition. When DCD drops, the Rx  
> switches to "listen for tune".  
>  
> Pawel  
>

73 All, Walt

## 18. Re: [HFSIG:453] Re: ANDVT TACTERM Documentation

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Wed, 12 Jul 95 23:51:24 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sWGFA-0002G1C@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

The 16 & 39 tone modems are described in U.S. MIL-STD-188-110A. I  
loaned out my copy but several individuals on the group have copies.

Walt

In your message of 12 Jul 1995 at 0633 CDT, you write:

> According to WALT DUBOSE - K5YFW:  
>  
> > I used the TACTERM on HF data at 2400 BPS (and voice) before and  
> > during Desert Shield/Storm and it beats the pants off of 110 baud (and  
> > 300 baud) ASCII, packet, AMTOR and the like...and that is without FEC.  
> > I can just imagine what the 16 tone modem would do with a little FEC.  
>  
> > This would be a good model to duplicate using DSP and a soundcard as  
> > a first effort in getting high speed data on HF.  
>  
> Have you any information about the technology used by this TACTERM  
> (hardware, software) ?  
>

> Claude  
>

---

## 19. Re: [HFSIG:455] Re: ANDVT TACTERM Documentation

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Thu, 13 Jul 1995 09:23:07 +0200 (MET DST)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.950713085413.29937A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Wed, 12 Jul 1995 JALOCHA@chopin.ifj.edu.pl wrote:

>  
> Did they say where are the audio band edges ?  
> Why don't they make a rig with 1 kHz bandwidth ? :-)  
> Actually it's easier to make a narrow crystal filter than a wide one.  
> But maybe the IC-738 filter's width is still switchable ?

The IF/audio bandwidth measured on a IC-736 transceiver in the ARRL lab is as follows at the - 6 dB points:

USB: 401 - 2102 Hz (1701 Hz)  
LSB: 431 - 2148 Hz (1717 Hz)

So as you can see rather narrow. The radio has passband tuning and a position for fitting a second smaller filter. Usually a 500 Hz filter is fitted in the two IFs (9 MHz and 455 MHz).

On the other hand most of the military radios have an IF/audio bandwidth of 300 - 3000 Hz at the -3db points, with a group delay variation of +-0.5 ms between 800 - 2800 Hz. I think no HAM transceiver can match this.

>  
>By the way is there an optimal symbol rate for HF and DQPSK ?  
> If too high, the multi-paths come into effect,  
> if too low, the phase incoherency affects the data...  
> so where is the optimum ?

Pawel is always asking difficult questions, :-)  
It is clear there is an optimum rate at a given moment and for given propagation conditions, the only problem is that the propagation varies all the time. I have seen a study made by the Swedish Defence establishment that tested FSK modulation rates between 37 Baud and 600 Baud. They found that you even could run happily at 600 Bauds from time to time, specially during day time on a good day time frequency. One thing to keep in mind when reading such studies is that the military usually work with links between 100 - 1000 km, so there is no real DX involved, :-)

>From what I have read so far I think 75 Baud is a very conservative rate which undoubtedly will work under almost all conditions. Personally I think the optimal, safe rate is closer to 125 Baud. As far as I have understood it should also be safer to run 125 Baud DQPSK than 125 Baud FSK, but this might again depend on the design of the FSK demodulator.

I have seen a reference to one modem using 66 tones with 37.5 Baud. I can't quite remember what waveform it used, but I think it was DPSK. Personally I think I would use less tones and a higher symbol rate.

73's Frode

|                                |        |                        |
|--------------------------------|--------|------------------------|
| Frode Weierud                  | Phone  | : 41 22 7674794        |
| CERN, SL                       | Fax    | : 41 22 7679185        |
| CH-1211 Geneva 23, Switzerland | E-mail | : frode@dxcern.cern.ch |

---

## 20. address

From: SONNTAG@vaxa.acdnj.itt.com  
Date: 13 Jul 95 10:33:00 EST  
To: "hfsig" <hfsig@tapr.org>  
Message-ID: <199507131438.JAA10865@dingus.n5lyt.datarace.com>  
Rendered body type: text/plain

For some reason my address has spontaneously changed to an incorrect address. Please change "Snntag@..." to "Sonntag@...".

Thanks!

---

## 21. Re: [HFSIG:459] address

From: Greg Jones <gjones@tenet.edu>  
Date: Thu, 13 Jul 1995 10:40:36 -0500 (CDT)  
To: hfsig@tapr.org  
Cc: n5lyt@dingus.n5lyt.datarace.com (Lee Ziegenhals)  
Message-ID: <199507131540.KAA01205@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

I am sorry, but on the listserv you are listed as: SONNTAG@VAXA.ACDNJ.ITT.COM

Your problem might be elsewhere.

Cheers - Greg

According to SONNTAG@vaxa.acdnj.itt.com:

>  
> For some reason my address has spontaneously changed to an incorrect address. Please  
> change "Snntag@..." to "Sonntag@...".  
>  
> Thanks!  
>  
>

---

## 22. Hilbert

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Thu, 13 Jul 1995 17:21:51 +0100  
To: hfsig@tapr.org  
Message-ID: <199507131812.AA24625@felix.dircon.co.uk>  
Rendered body type: text/plain

Here is a question for the group,

If one implements a Hilbert Transform using a FIR filter and the Parks Mc Clelland algorithm which sample in the input buffer is the one that is 90 deg phase different? is it the one in the middle?

Regards Confused!

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 23. Re: [HFSIG:458] Re: ANDVT TACTERM Documentation

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Thu, 13 Jul 95 18:18:39 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sWXWh-0002FEC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

In Frode's message of 13 Jul 1995 at 0230 CDT, he writes:

> On Wed, 12 Jul 1995 JALOCHA@chopin.ifj.edu.pl wrote:  
>  
> >  
> > Did they say where are the audio band edges ?  
> > Why don't they make a rig with 1 kHz bandwidth ? :-)  
> > Actually it's easier to make a narrow crystal filter than a wide one.  
> > But maybe the IC-738 filter's width is still switchable ?  
>  
> The IF/audio bandwidth measured on a IC-736 transceiver in the ARRL lab  
> is as follows at the - 6 dB points:  
>     USB: 401 - 2102 Hz (1701 Hz)  
>     LSB: 431 - 2148 Hz (1717 Hz)  
>  
> So as you can see rather narrow. The radio has passband tuning and a  
> position for fitting a second smaller filter. Usually a 500 Hz filter is  
> fitted in the two IFs (9 MHz and 455 MHz).  
>  
> On the other hand most of the military radios have an IF/audio bandwidth  
> of 300 - 3000 Hz at the -3db points, with a group delay variation of  
> +-0.5 ms between 800 - 2800 Hz. I think no HAM transceiver can match this.

The Harris transceivers used by the U.S. DoD (AN/URC-119) do indeed have a 300 - 3000 Hz, 3db bandpass as well as others used by DoD agencies. This is one reason the TACTERM and other MIL-STD-188-110A modems work so well with these systems.

IMHO, we (hams) will have to decide if we want to "build" a modem to fit into a 500 - 2100 Hz -6db bandpass or replace the filters in our HF rigs for ones that will be 300 - 3000 Hz.

Admittedly those of us interested in high speed HF data transmission will initially be a very small minority. However, I believe once we can show that robust 2400 BPS data can be sent on HF, there will be many clamoring to get this "new" ham technology. At this point, manufacturers may begin offering an optional wide SSB filter.

I would recommend that you (I didn't say we because I'm not doing the "work") begin with a modulation scheme that will fit into a 500 - 2100 Hz bandpass with the ability to expand to a scheme that can use a 300 - 3000 Hz bandpass.

- > Pawel is always asking difficult questions, :-)
- > It is clear there is an optimum rate at a given moment and for given
- > propagation conditions, the only problem is that the propagation varies
- > all the time. I have seen a study made by the Swedish Defence
- > establishment that tested FSK modulation rates between 37 Baud and 600 Baud.
- > They found that you even could run happily at 600 Bauds from time to
- > time, specially during day time on a good day time frequency. One thing
- > to keep in mind when reading such studies is that the military usually
- > work with links between 100 - 1000 km, so there is no real DX involved, :-)

The above is mostly true...actually 40/45 - 1200/1500 km and they try to punch thru as much data as possible in the shortest time. Thus, they will use 300 (even 600) baud AFKS when conditions would hardly support 150 baud.

- > >From what I have read so far I think 75 Baud is a very conservative rate
- > which undoubtedly will work under almost all conditions. Personally I
- > think the optimal, safe rate is closer to 125 Baud. As far as I have
- > understood it should also be safer to run 125 Baud DQPSK than 125 Baud
- > FSK, but this might again depend on the design of the FSK demodulator.

HAL, in their CLOVER II papers, suggest that 150 baud is the maximum baud rate to consider using on HF. This is also suggested by Harris Corp, Rockwell Collins and Magnavox. They may all be using data from research done by Stanford Research Institute. Many "old-timers" say the reason RTTY and ASCII were never used above 100/110 baud, respectively, is that at the end of WWII it was determined that these were the maximum baud rates that HF would support.

- > I have seen a reference to one modem using 66 tones with 37.5 Baud. I
- > can't quite remember what waveform it used, but I think it was DPSK.
- > Personal ly I think I
- > would use less tones and a higher symbol rate. >
- > 73's Frode
- >
- >
- > Frode Weierud Phone : 41 22 7674794
- > CERN, SL Fax : 41 22 7679185
- > CH-1211 Geneva 23, Switzerland E-mail : frode@dxcern.cern.ch
- >

73, Walt@home.2.nite

=====  
Kelly AFB, TX and McClelland AFB, CA were \*not\* saved. Another great U.S. military mistake. We should have learned from General Custer's mistake.

This is my personal opinion and does in no way represent the view of the DoD, USAF or any other official U.S. Government Agency.

Walter D. DuBose

=====

---

## 24. Re: [HFSIG:461] Hilbert

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Thu, 13 Jul 1995 15:53:30 -0800  
To: hfsig@tapr.org  
Message-ID: <9507132341.AA04815@ucs.orst.edu>  
Rendered body type: text/plain

Charles,

Good question.

>Here is a question for the group,  
>  
> If one implements a Hilbert Transform using a FIR filter and the  
> Parks-McClelland algorithm which sample in the input buffer is the  
> one that is 90 deg phase different? is it the one in the middle?  
>

Lets see - think of it this way. Consider the Hilbert transform as a black box where you feed your real-valued time series in. The Hilbert transformer takes one input and has two outputs, the two outputs are identical except that they are 90 degrees out of phase, i.e. like sine is the same as a cosine 90 degrees phase shifted. The output of the Hilbert transformer is now an analytical signal where one branch could be thought of as the "real" and the other the "imaginary". This arrangement has some "magical" properties when it comes to modulation/demodulation but I'll leave that for the moment. To return to the question: EVERY point of the input buffer is affected by the transformation. Its kind of similar as doing an FFT and taking a frequency bin and asking "which data point from the time series made this bin's content?" The answer is "each and every data point".

Some other useful tips about the Hilbert transform.

-----

- 1) Be aware of Parks-McClellan FIR code laying around the Internet. Many version does not compute a valid Hilbert transform - the way you know is to inspect the coefficients, they should alternate with every other one being equal zero.
- 2) If you use a single FIR Hilbert transform for your "Q" branch, make sure that the other branch i.e. your "I" is delayed by the "same" group delay, i.e. if your Hilbert transform has 39 taps, you need a delay of  $39/2 = 19.5$  (use either 19 or 20) taps delay for your "I".
- 3) One may also do the same thing using regular "comb" FIR filters (every other coefficient being zero). Then derive two versions from this set of coefficients, one multiplied by cosine, the other by sine. Now you will have 90 degree difference in the paths, but also have the correct group delay in each path.
- 4) If you use the P-McC algorithm for designing the transform, you will find that it is impossible to attain infinitely narrow transition zones. These will result in "holes" in the transform - particularly at the origin and band edge. Most often you may just ignore these effects, but be aware that you need high enough order filter (look at the number of dB rejection you get - anything better than say 30-40 dB is usable).

Above is more or less correct - feel free to jump in and correct me otherwise. I'm also just learning about the finer points.

73's

--Johan

---

## 25. Re: [HFSIG:455] Re: ANDVT TACTERM Documentation

From: bm@lynx.ve3jf.ampr.org (Barry McLarnon VE3JF)  
Date: Fri, 14 Jul 1995 02:24:56 +0000 (GMT)  
To: hfsig@tapr.org  
Message-ID: <m0sWaQy-0002qeC@lynx.ve3jf.ampr.org>  
Rendered body type: text/plain

Pawel asks:

- > By the way is there an optimal symbol rate for HF and DQPSK ?
- > If too high, the multi-paths come into effect,
- > if too low, the phase incoherency affects the data...
- > so where is the optimum ?

This is an age-old question. :-) I have a copy of a paper from 1962 (8th Nat'l Communications Symposium) by one Heinz Fiege-Kollmann, entitled "The optimum bit length for HF data transmission", in which he concludes that the optimum symbol length for DPSK lies somewhere between 5 ms and 15 ms. He says that the optimum is a broad one, but that

performance degrades more quickly as the symbol length is shortened than when it is increased. This was based on work done during the development of the granddaddy of all parallel-tone modems, the Collins Kineplex, in the mid- to late-50's. The oldtimers usually used 75 or 100 baud, and they probably knew what they were doing... :-)

Just think of all those racks of equipment that can now be replaced by one DSP chip...

One common technique in parallel-tone modems that I don't recall seeing mentioned here (but I probably missed it) is the use of a guard interval during the first part of the symbol interval. Energy received during the guard interval, which hopefully includes most of the multipath, is excluded from the decision process. One set of parameters that comes to mind (probably from Kineplex) is an overall symbol length of 13.33 ms and a useful symbol length of 9.09 ms, giving 4.24 ms of guard time and an orthogonal tone spacing of 110 Hz. Guard intervals are also used for multipath mitigation in the more recent high-speed VHF/UHF OFDM systems, such as the Eureka 147 DAB system. For example, Eureka Mode 1 has a 0.25 ms guard interval, 1 ms useful symbol length, and 1536 tones(!) with 1 kHz spacing.

Barry

--

Barry McLarnon VE3JF/VA3TCP  
Ottawa Amateur Radio Club Packet Working Group  
Email: bm@hydra.carleton.ca or bm@lynx.ve3jf.ampr.org

---

## 26. Re: [HFSIG:464] Re: ANDVT TACTERM Documentation

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Thu, 13 Jul 1995 22:13:33 -0800  
To: hfsig@tapr.org  
Message-ID: <9507140601.AA12819@ucs.orst.edu>  
Rendered body type: text/plain

Hi Barry,

<<---some lines deleted--->>

>One common technique in parallel-tone modems that I don't recall seeing  
>mentioned here (but I probably missed it) is the use of a guard interval  
>during the first part of the symbol interval. Energy received during  
>the guard interval, which hopefully includes most of the multipath, is  
>excluded from the decision process. One set of parameters that comes to  
>mind (probably from Kineplex) is an overall symbol length of 13.33 ms  
>and a useful symbol length of 9.09 ms, giving 4.24 ms of guard time and  
>an orthogonal tone spacing of 110 Hz. Guard intervals are also used for  
>multipath mitigation in the more recent high-speed VHF/UHF OFDM systems,  
>such as the Eureka 147 DAB system. For example, Eureka Mode 1 has a 0.25  
>ms guard interval, 1 ms useful symbol length, and 1536 tones(!) with 1  
>kHz spacing.

Yes, quite right - thanks for reminding us about this issue:

In addition to the multipath performance, there is yet another reason for the guard bands:

In J.D. Ralphs' book: Principles and Practice of Multi-frequency Telegraphy, page 41, the theory of the guard time is discussed. The gist of the idea is that most of the spectral spill-over into adjacent bins occurs at bit transitions and are mostly noticeable right right at or near those time instances. If a "dead" zone or guard interval is used (i.e. the integrator/FFT excluding that part of the signal), he shows a significant improvement in demodulator S/N is possible due to using the guard band.

This same book (page 113) looks at the 16 DPSK parallel-tone modem and mentions a guard band of 4 ms out of the 13.3ms (75 baud) used for those modems. There are some further examples of this topic - I'm too lazy to look for them at the moment.

--Johan Forrer, KC7WW

---

## 27. Re: [HFSIG:463] Re: Hilbert

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Fri, 14 Jul 1995 06:03:17 +0100



To: hfsig@tapr.org  
Message-ID: <199507140653.AA17004@felix.dircon.co.uk>  
Rendered body type: text/plain

>2) If you use a single FIR Hilbert transform for your "Q" branch, make sure  
>that the other branch i.e. your "I" is delayed by the "same" group delay,  
>i.e. if your Hilbert transform has 39 taps, you need a delay of  $39/2 = 19.5$   
>(use either 19 or 20) taps delay for your "I".  
>

>73's

>  
>--Johan  
>

Thanks Johan,

The above is what I wanted to know!

I am using a program called PC-DSP written by Oktay Alkin, I have used it to design FIR filters. It is the program that Jon Bloom KE3Z described in QEX a year ago (it comes with a book and costs \$29).

Amongst other things it allows you to design a Hilbert transform either using Parks  
Mc Clelland or Fourier.

I was thinking of incorporating it into my 8 ary modem on the C50. However it seems to have enough processing power to work with a larger complex transform. I guess I could use Goertzel (???) algorithm instead!

Regards Charles

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 28. Re: [HFSIG:462] Re: ANDVT TACTERM Documentation

From: Frode Weierud <frode@dx.cern.ch>  
Date: Fri, 14 Jul 1995 10:44:22 +0200 (MET DST)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.950714094315.20933A-100000@dx.cern.ch>  
Rendered body type: text/plain

On Thu, 13 Jul 1995, WALT DUBOSE - K5YFW wrote:

>  
> IMHO, we (hams) will have to decide if we want to "build" a  
> modem to fit into a 500 -2100 Hz -6db bandpass or replace the  
> filters in our HF rigs for ones that will be 300 - 3000 Hz.  
>  
> Admittedly those of us interested in high speed HF data  
> transmission will initially be a very small minority. However,  
> I believe once we can show that robust 2400 BPS data can be  
> sent on HF, there will be many clamoring to get this "new"  
> ham technology. At this point, manufacturers may begin  
> offering an optional wide SSB filter.  
>  
> I would recommend that you (I didn't say we because I'm not  
> doing the "work") begin with a modulation scheme that will  
> fit into a 500 - 2100 Hz bandpass with the ability to expand  
> to a scheme that can use a 300 - 3000 Hz bandpass.

I agree with you that this is the most promising approach. We should reduce the number of tones to be able to nicely fit within the available bandwidth of standard ham transceivers. We should make the modem such that it would be easy to reconfigure it to work with different number of tones. However to have something which is generally useable, it would mean that we will have to have a way of distinguishing between a transmissions using the full tone set and ones using a reduced tone set. Perhaps this could be done by using a different number of Doppler tones.

It is rather clear from recent papers that there is trend away from the parallel tone modems and towards the single tone modems. I read recently a comparison between these types of modems and it was indicated that you would need 15 dB more power with a parallel modem than with the serial tone modem to achieve an error rate of  $1E-3$ . During this comparison both modems were running without ECC. I nevertheless think we are doing the right thing in going for the parallel tone modems for the

time being as the serial tone modems need rather complex channel equalizers and channel quality analysis algorithms. Serial tone modems should come when we have got thorough experience with the parallel tone ones.

The ANDVT TACTERM modem is using an EOM (End-Of-Message) sequence that was meant to insure a rapid changeover between receive/transmit. Initially I thought this would have little use for amateur use, but I am now wondering if it would be useful after all. It would insure that at the end of transmission the receiving station would quickly be able to take the link, or to fall back into preamble tracking mode. If not used it is possible that an error burst could confuse the receiver and that it would stay in data receive mode, while the transmitting station is sending a new Doppler/Sync sequence.

Would this be of any use? Anyone with an opinion?

73's Frode

|                                |        |                        |
|--------------------------------|--------|------------------------|
| Frode Weierud                  | Phone  | : 41 22 7674794        |
| CERN, SL                       | Fax    | : 41 22 7679185        |
| CH-1211 Geneva 23, Switzerland | E-mail | : frode@dxcern.cern.ch |

---

## 29. Re: [HFSIG:467] Re: ANDVT TACTERM Documentation

From: JALOCHA@chopin.ifj.edu.pl  
Date: Fri, 14 Jul 1995 13:58 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <7A637F4DE022C218@chopin.ifj.edu.pl>  
Rendered body type: text/plain

> IMHO, we (hams) will have to decide if we want to "build" a  
> modem to fit into a 500 -2100 Hz -6db bandpass or replace the  
> filters in our HF rigs for ones that will be 300 - 3000 Hz.

My two cents:

cent #1:

I want to stress that for the multi-tone modem the passband uniformity is not very much critical. It should accept ripples up to 12-18 dB (the Tx filter is in must more critical). This is because every tone occupies small bandwidth and it is being decoded independently. That if carrier #1 has an amplitude of 10 but carrier #8 only 1 it really doesn't matter that much.  
What I see for the FT757GXII is that the SSB filter passes 300-3000 Hz within maybe 6dB - this you can see with your S-meter while tuning across a dead carrier (turn on the internal calibrator). Then the post-demodulation audio filters do the "nasty" job of attenuation the higher frequencies so on the output the last carrier is 3 times weaker than the "top" one. But here I don't really care... the noise gets 3 times weaker too, so the demodulator will experience as many errors as for any other carrier.

cent #2:

When I was younger :-) I read many book on building SSB transceivers and all the SSB filters mentioned there were 2.7 kHz wide - looked to me like a very standard bandwidth.

> Admittedly those of us interested in high speed HF data  
> transmission will initially be a very small minority. However,  
> I believe once we can show that robust 2400 BPS data can be  
> sent on HF, there will be many clamoring to get this "new"  
> ham technology. At this point, manufacturers may begin  
> offering an optional wide SSB filter.

They should just offer a standard 2.7 kHz filter - no need for "wide" :-)  
What might be nice however, is the capability to bypass all the pre-modulation and post-demodulation audio circuits.

>It is rather clear from recent papers that there is trend away from  
>the parallel tone modems and towards the single tone modems.

We should learn the "single tone modem" principles: does it use some sort of equalizer to remove the effect of multipaths ?

>I read

>recently a comparison between these types of modems and it was indicated  
>that you would need 15 dB more power with a parallel modem than with the  
>serial tone modem to achieve an error rate of 1E-3.

And what is the reason for such a difference ?  
This effect may come from the fact that for a multi-tone modem  
you need to transmit with lower average power because the envelope  
of the signal is not constant. So was the "power" the transceiver power  
or was this the signal's power on the air ?

>The ANDVT TACTERM modem is using an EOM (End-Of-Message) sequence that  
>was meant to insure a rapid changeover between receive/transmit. Initially  
>I thought this would have little use for amateur use, but I am now  
>wondering if it would be useful after all.

As for now I am going to use EOM sequence but different that the one for  
TACTERM. I will transmit random data with the maximum phase error  
(45 degrees) with the intention to generate maximum receiver decoder  
errors. I hope this will speed up the DCD switch off. If a noise burst  
comes right at the EOM, it will only slow down the DCD switch off by  
a factor of two.

My code for the DSPCARD4 acquires proper symbol sync. now, and switches  
to the data decoding phase. It even decodes the data I only didn't check  
yet if the data come out right.

Pawel

---

### 30. Re: [HFSIG:468] Re: ANDVT TACTERM Documentation

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Fri, 14 Jul 1995 17:09:23 +0200 (MET DST)  
To: hfsig@tapr.org  
Message-ID: <Pine.Ult.3.91.950714162737.7711A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Fri, 14 Jul 1995 JALOCHA@chopin.ifj.edu.pl wrote:

>  
> cent #2:  
> When I was younger ;-) I read many book on building SSB trancievers  
> and all the SSB filters mentioned there were 2.7 kHz wide - looked to me  
> like a very standard bandwidth.

You are right that was the standard, but with the band conditions getting  
worse there has been a recent trend to make filters with 1.8 to 2.1 kHz  
bandwidth. I think the 1.8 kHz filters are really for the contest  
operators, but more and more rigs appear with the narrow type of SSB  
filter at 2.1 kHz as the standard filter. Pawel, you are getting old ;-)

>  
> They should just offer a standard 2.7 kHz filter - no need for "wide" :-)  
> What might be nice however, is the capability to bypass all the pre-modulation  
> and post-demodulation audio circuits.

I think this is not a bad idea. It should be possible to switch out the  
audio shaping when running the type of digital modes we are talking about.

>  
> We should learn the "single tone modem" principles: does it use some sort  
> of equalizer to remove the effect of multipaths ?

Yes, they are using equalizers, but not simple static equalizers. The  
equalizers are adaptive to be able to cope with the randomly changing HF  
channel. They therefore are using RTCE (Real Time Channel Estimation) to  
evaluate the channel and drive the adaptive equalizer. This can be rather  
complex. Some serial tone modems are using special training sequences  
after the preamble which is then analyzed and evaluted by the receiving  
modem. These modems today run around 3000 bps in a 1800 Hz bandwidth,  
QDPSK modulating a single carrier usually at 1800 Hz. The Ecotel modem  
from Telefunken I think is using 1650 Hz as the carrier frequency.

>  
>>I read  
>>recently a comparison between these types of modems and it was indicated  
>>that you would need 15 dB more power with a parallel modem than with the  
>>serial tone modem to achieve an error rate of 1E-3.  
>  
> And what is the reason for such a difference ?  
> This effect may come from the fact that for a multi-tone modem  
> you need to transmit with lower average power because the envelope  
> of the signal is not constant. So was the "power" the transceiver power  
> or was this the signal's power on the air ?  
>

This I got from their published S/N ratio versus BER (bit-error rate) diagram that showed about 15dB lower S/N figure for the serial tone modem for a BER of 1E-3. So I sort of extrapolated that to mean that I would need a 15dB stronger signal from the parallel tone modem to get the same BER figure, or did I interpret that wrong?

>  
> As for now I am going to use EOM sequence but different that the one for  
> TACTERM. I will transmit random data with the maximum phase error  
> (45 degrees) with the intention to generate maximum receiver decoder  
> errors. I hope this will speed up the DCD switch off. If a noise burst  
> comes right at the EOM, it will only slow down the DCD switch off by  
> a factor of two.

That agrees with my idea as well. Your method is perhaps a lot easier than to try to correlate a PN sequence to detect the EOM. If it does the same job correctly why not.

>  
> My code for the DSPCARD4 acquires proper symbol sync. now, and switches  
> to the data decoding phase. It even decodes the data I only didn't check  
> yet if the data come out right.  
>

I have even heard some rumours that your waveform already have made it on the air via HF. I hope we will get some more reports on these trials.

73's Frode

|                                |        |                        |
|--------------------------------|--------|------------------------|
| Frode Weierud                  | Phone  | : 41 22 7674794        |
| CERN, SL                       | Fax    | : 41 22 7679185        |
| CH-1211 Geneva 23, Switzerland | E-mail | : frode@dxcern.cern.ch |

---

### 31. Re: [HFSIG:466] Re: Hilbert

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 14 Jul 1995 09:36:27 -0800  
To: hfsig@tapr.org  
Message-ID: <9507141724.AA15631@ucs.orst.edu>  
Rendered body type: text/plain

Charles,

> I am using a program called PC-DSP written by Oktay Alkin, I have used it to  
> design FIR filters. It is the program that Jon Bloom KE3Z described in QEX a  
> year ago (it comes with a book and costs \$29).  
> Amongst other things it allows you to design a Hilbert transform either  
> using Parks  
> Mc Clelland or Fourier.

Sounds like you have it all under control.

> I was thinking of incorporating it into my 8 ary modem on the C50. However  
> it seems to have enough processing power to work with a larger complex  
> transform. I guess I could use Goertzel (???) algorithm instead!  
>

If I may add my two cents' worth:

Interesting that you should bring this up. Goertzel's method is nice if you only need a few FFT bins and probably will be OK for what you need. However, think about this for a minute: When you compare the spectral energy in a particular bin as measured with an FFT vs. what you get when you use a nice sharp FIR filter, you will find that the dynamic range, i.e. the magnitude of your biggest - smallest number, is much smaller for the FFT case, than for the FIR output. It is not difficult to see why this is the case when one considers that a frequency bin is made up of many summed branches, all containing multiplies. A FIR filter, on the other hand, has a very predictable outcome as far as saturation and dynamic range is concerned, especially if coefficients have been engineered to sum to "unity" response in the worst case scenario. Why folks don't use them, is because you need to many taps to make a decent filter bank - DSP's usually run out of clock ticks. (There are tricks to lower the filter order, but at a cost of much higher sampling rates. I'll leave it at that for the moment.)

That being the case, and things getting even worse for an integer FFT, you may want to consider doing the floating point Goertzel. I have seen the floating point Goertzel in some books. I believe that the 'C50 has some special instructions that helps out doing floating point? This should already help a lot for weak signals.

Good luck with your project.

--Johan, KC7WW

---

## 32. Re: [HFSIG:467] Re: ANDVT TACTERM Documentation

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 14 Jul 1995 09:36:37 -0800  
To: hfsig@tapr.org  
Message-ID: <9507141724.AA15857@ucs.orst.edu>  
Rendered body type: text/plain

>On Thu, 13 Jul 1995, WALT DUBOSE - K5YFW wrote:  
>  
>>  
>> IMHO, we (hams) will have to decide if we want to "build" a  
>> modem to fit into a 500 -2100 Hz -6db bandpass or replace the  
>> filters in our HF rigs for ones that will be 300 - 3000 Hz.  
>>  
>> Admittedly those of us interested in high speed HF data  
>> transmission will initially be a very small minority. However,  
>> I believe once we can show that robust 2400 BPS data can be  
>> sent on HF, there will be many clamoring to get this "new"  
>> ham technology. At this point, manufacturers may begin  
>> offering an optional wide SSB filter.  
>>  
>> I would recommend that you (I didn't say we because I'm not  
>> doing the "work") begin with a modulation scheme that will  
>> fit into a 500 - 2100 Hz bandpass with the ability to expand  
>> to a scheme that can use a 300 - 3000 Hz bandpass.  
>

In a document that I received from Phil Karn, it describes in a fair amount of detail, how one outfit did their own 16-tone MIL-STD-188 modem. They also had to deal with this same problem of using readily-available SSB rigs. Their solution was to use an audio band 935 - 2585 Hz for the tones.

>I agree with you that this is the most promising approach. We should  
>reduce the number of tones to be able to nicely fit within the available  
>bandwidth of standard ham transceivers. We should make the modem such that  
>it would be easy to reconfigure it to work with different number of  
>tones. However to have something which is generally useable, it would  
>mean that we will have to have a way of distinguishing between a  
>transmissions using the full tone set and ones using a reduced tone set.  
>Perhaps this could be done by using a different number of Doppler tones.  
>

>It is rather clear from recent papers that there is trend away from  
>the parallel tone modems and towards the single tone modems. I read  
>recently a comparison between these types of modems and it was indicated  
>that you would need 15 dB more power with a parallel modem than with the  
>serial tone modem to achieve an error rate of 1E-3. During this  
>comparison both modems were running without ECC. I nevertheless think we  
>are doing the right thing in going for the parallel tone modems for the  
>time being as the serial tone modems need rather complex channel  
>equalizers and channel quality analysis algorithms. Serial tone modems  
>should come when we have got thorough experience with the parallel tone  
>ones.

I also believe that there is no question that a single tone system at low baud rate will outperform a parallel tone system (in terms of BER for any given S/N). However, in the end it becomes a tradeoff between speed and robustness. The parallel modem will certainly have the advantage for speed. I think one have to be careful when reading papers about these things without normalizing everything to the same units (BER for given S/N at bits/second/hertz). This is confusing.

>  
>The ANDVT TACTERM modem is using an EOM (End-Of-Message) sequence that  
>was meant to insure a rapid changeover between receive/transmit. Initially  
>I though this would have little use for amateur use, but I am now  
>wondering if it would be useful after all. It would insure that at the

>end of transmission the receiving station would quickly be able to take  
>the link, or to fall back into preamble tracking mode. If not used it is  
>possible that an error burst could confuse the receiver and that it would  
>stay in data receive mode, while the transmitting station is sending a

In an FEC situation, especially where variable length messages are sent, every little bit of framing/blocking information is helpful to the remote decoder. In AMTOR FEC, for example, there is such a "go back to standby" sequence sent at the end of the transmission. It is repeated several times and only one of those is needed to get the remote to act on. If the remote missed it due to interference, then a rapid succession of errors tells that it has lost sync, whereupon it goes back to hunt for sync.

If the sync pre-amble is designed cleverly, it may be possible to write an algorithm that is hunting for it all the time - a daemon. When it sees it, it forces a frame reset. As an interesting aside, in Pactor, the designers choose such a poor sync header, that this is not possible - the remote decoder have to find frames by brute force search for a valid CRC, i.e. a whole frame buffer of samples have to be searched at every new data sample - then once the CRC is valid, can one inspect the sync symbol. This is one reason why the early Pactor units could not do automatic signal detection of all modes - there Z80 processor was out of cycles.

--Johan

---

### 33. DCC Paper Deadline!

From: Greg Jones <gjones@tenet.edu>  
Date: Fri, 14 Jul 1995 13:40:11 -0500 (CDT)  
To: tapr-bb@tapr.org (TAPR-BB mailing), netsig@tapr.org (NETSIG mailing),  
bbssig@tapr.org (BBS SIG mailing), hfsig@tapr.org (HF SIG mailing),  
dsp-93@tapr.org (DSP-93 Build), aprssig@tapr.org (BBS SIG mailing),  
amsat-bb@amsat.org (AMSAT BB Mail Group)  
Message-ID: <199507141840.NAA28968@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

Don't forget -- Friday, July 21st! Deadline for ARRL DCC Article Submissions  
-----

- \* Anyone interested in digital communications is invited to submit a paper for publication in the Conference Proceedings. Articles are welcome on any aspect of digital communications.
- \* Presentation at the Conference is not required for publication.
- \* Papers are due by July 21, 1995, and should be submitted to  
Maty Weinberg, ARRL, 225 Main Street, Newington, CT 06111 or via the  
Internet to mweinberg@arrl.org
- \* Please contact Maty for detailed format requirements.
- \* It is not too late to start writing!

---

If you think you will be late in submitting your paper(s), contact Maty to  
arrange details.

---

14th Annual ARRL Digital Communications Conference  
September 8-10th, 1995  
Arlington, Texas (minutes from DFW airport)

The ARRL Digital Communications Conference is an international forum for radio amateurs in digital communications, networking, and related technologies to meet, publish their work, and present new ideas and techniques for discussion. Presenters and attendees will have the opportunity to exchange ideas and learn about recent hardware and software advances, theories, experimental results, and practical applications. The Digital Communications Conference is not just for the digital elite, but for digitally-orientated amateurs of all levels of experience.

For further information:

E-Mail: TAPR@TAPR.ORG  
Web : <http://www.tapr.org/tapr>  
ftp : [ftp://ftp.tapr.org/tapr/95\\_DCC\\_Flyer.pdf](ftp://ftp.tapr.org/tapr/95_DCC_Flyer.pdf)

### 34. Re: [HFSIG:469] Re: ANDVT TACTERM Documentation

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Fri, 14 Jul 95 14:42:15 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sWqcq-0001v2C@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

The Thread sez:

> > They should just offer a standard 2.7 kHz filter - no need for "wide" :-)  
> > What might be nice however, is the capability to bypass all the pre-modulation  
> > and post-demodulation audio circuits.  
>  
> I think this is not a bad idea. It should be possible to switch out the  
> audio shaping when running the type of digital modes we are talking about.  
>

This is a reality check:

Getting a manufacturer to "custom" build an HF rig to fit the needs of a small groups of users (like us) is unlikely...Why? Because I dare say that few HF rig manufacturers build more than 50,000 of any certain model (at an average of \$600US, that's \$30,000,000 in wholesale sales).

It is more than likely that they are building HF rigs with the filters they have to support the large DX community where a narrow bandpass is desirable.

I believe you must design a modem that can be used with the HF rigs on the market (today) and prove the modems worth before you will find manufactures building and selling HF rigs with the filtering/audio response you/we desire

> I have even heard some rumors that your waveform already have made it on  
> the air via HF. I hope we will get some more reports on these trials.  
>

If you mean 39 or 16 parallel tone modems (such as in the TACTERM), there are many U.S. DoD agencies and some commercial HF users carrying on HF data communications using these modem protocols. I have heard what sounds like strong noise peaks on HF and suspect that many of these are the multi-tone and serial tone modems at work.

73, Walt

---

### 35. CLOVER II vs. Pactor II

From: willi\_r@mail.uwlax.edu  
Date: Fri, 14 Jul 95 19:01:55 -0500  
To: hfsig@tapr.org  
Message-ID: <9507150001.AA10355@mail.uwlax.edu>  
Rendered body type: text/plain

There were two things holding me back from making a decision to "invest" in new hf modem hardware. 1) Legal operation of automatic and semi-automatic operation in the U.S. and 2) Which is the best hf mode for hi speed transfer under all condx whether for highest speed or best thruput under deteriorating condx.

Well #1 is now settled and legal here in U.S.A.

But #2 is still up in the air.

Anybody have any preliminary information on what the comparison will be? OK, how about any opinions? Pactor II sounds better than CLOVER II in that it may have faster thruput and better weak signal abilities, but we need some tests. Surely, some of you folks who are the worlds top leaders in hf issues are either doing some tests yourself or know of others. This would be bugging me to no end if I had the equipment. I would have to know.:-)

The alternative is for one of the digital ham companies to come out with a unit that will do both modes or better yet, do one now and have a guaranteed

upgrade path for a reasonable price (like no more than 100 bux) to the other mode within a year or so.

I have no idea of the approximate cost there is to license these modes, but if there was a DSP modem that had the horsepower and the support of the 3rd party software authors to make it work with Winlink and similar connectivity, I know I would be a buyer at the 500-700 bracket.

I did ask HAL about this but they said (in a very nice way) that they could not commit to this concept. Maybe licensing is really high priced but find it hard to believe it could be over 20-30 bux per license. Am I living in fantasyland?

Wonder if the TAPR DSP-93 board could handle these new modes? Wonder if it could get the support like the HAL boards do? I sure would not have a problem with paying a modest price for embedded call software for TAPR software (50 bux or so) similar to what the IDRA (International Digital Radio Association) formerly the ADRS does with their rather successful software.

Rick, KV9U

---

## 36. Re: [HFSIG:473] Re: ANDVT TACTERM Documentation

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Sat, 15 Jul 1995 17:41:02 +0200 (MET DST)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.950715173400.22055B-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Fri, 14 Jul 1995, WALT DUBOSE - K5YFW wrote:

>  
> I believe you must design a modem that can be used with the HF  
> rigs on the market (today) and prove the modems worth before  
> you will find manufactures building and selling HF rigs with  
> the filtering/audio response you/we desire

I agree we will probably not get the rig manufacturers to tailor their rigs to our needs. What I had in mind was simply a data input/output like you now find on the new VHF/UHF rigs for packet radio. This data I/O will simply bypass the audio shaping that you have in most rig. It will not solve the problem with lousy filters, but it should improve things slightly.

> > I have even heard some rumors that your waveform already have made it on  
> > the air via HF. I hope we will get some more reports on these trials.  
> >

>  
> If you mean 39 or 16 parallel tone modems (such as in the  
> TACTERM), there are many U.S. DoD agencies and some  
> commercial HF users carrying on HF data communications using  
> these modem protocols. I have heard what sounds like strong  
> noise peaks on HF and suspect that many of these are the  
> multi-tone and serial tone modems at work.  
>

Yes, I have heard several of these transmissions. They are on every day. What I really meant with my remark is that Pawel's multitone modem has made it on HF. A few hams are now actively testing his modem on VHF and HF. Just to let you see that things are moving extremely fast ahead.

73's Frode

|                                |        |                        |
|--------------------------------|--------|------------------------|
| Frode Weierud                  | Phone  | : 41 22 7674794        |
| CERN, SL                       | Fax    | : 41 22 7679185        |
| CH-1211 Geneva 23, Switzerland | E-mail | : frode@dxcern.cern.ch |

---

## 37. Parallel vs Serial Tone Modems

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Sun, 16 Jul 95 18:52:05 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sXdTi-00016QC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

Having used both parallel and serial tone MIL-STD-188-110A modems, I must admit that the serial tone modem appeared to be more robust. However, I really wonder if we (hams) need that robustness. We're talking about signals way down in the noise...channelized communications with known stations on the channel. I'm not sure this is really needed on the hambands. It would come in handy during the midst of a hurricane where the noise level is very high; but, we generally use



slow speed CW (5-10 wpm) done here on the Texas Gulf Coast.

I do not mean to stifle efforts in developing a serial tone modem...  
I'm just trying to be a little practical. As we say at Kelly AFB,  
"why buy a Cadillac when a Chevrolet will do".

How about both...then we can "have our cake and eat it too".

73, Walt

---

### 38. Re: [HFSIG:475] Re: ANDVT TACTERM Documentation

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Sun, 16 Jul 95 18:57:27 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sXdYu-0001m6C@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

In Frode's message of 15 Jul 1995 at 1045 CDT, he writes:  
> What I really meant with my remark is that Pawel's multitone modem has  
> made it on HF. A few hams are now actively testing his modem on VHF and  
> HF. Just to let you see that things are moving extremely fast ahead.  
>  
> 73's Frode

In the local youth vernacular.....C o o l ! -- wdd

---

### 39. Re: [HFSIG:476] Parallel vs Serial Tone Modems

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Mon, 17 Jul 1995 05:40:36 +0100  
To: hfsig@tapr.org  
Message-ID: <199507170631.AA13339@felix.dircon.co.uk>  
Rendered body type: text/plain

>I do not mean to stifle efforts in developing a serial tone modem...  
>I'm just trying to be a little practical.  
>73, Walt  
>  
>

Hey,

Walt we do things because they are there!!! Seriously I have been waiting for  
a discussion on serial tone modems. A simple slow speed modem appears to  
be quite do-able. The main problem is I think is the probable requirement  
to use  
multiple DSPs.

I have been reading about the equalisers used in mobile radio modems, I guess  
the real difference is the actual multipath delays are much greater  
therefore longer  
equalisers are needed for H.F.

Whether we use a training sequence or decision directed I cannot guess.  
However it may well be that a slow < 1kbit/s modem is more suitable for the ham  
environment especially something that is proof against those pesky CWers!

Something like a 300 baud modem would be interesting with a simple equaliser  
in front of it.

As far as receivers are concerned a nice project would be to take an I.F signal  
down convert it to say 10Khz then demodulate using DSP. I will have to  
investigate my TS850 as it does something like that for it's external DSP unit.  
(Although it just filters).

Another thing would be an intelligent VOX with compression done in DSP  
could event have an expander at the far end.

I must admit I have learn't a lot the last few months from this news group, my  
8ary modem is proceeding it has been re-written a couple of times due to  
my improving knowledge. My main problem currently is gaining and tracking  
bit sync. The sync has to lock as fast as possible. It should also have the  
ability to re-sync to a colliding signal this of course causes a problem!  
The technique I am using is to divide the tone with the highest power with  
that of the others. This produces a nice triangle wave. I then use an early/late  
gate to find the peak. This locks the NCO (thats the bit I can't get to work  
yet)  
the NCO then triggers an integrate and dump. At the end of the period a decision  
is made as to which tone was received.

Regards Charles

-- -----

"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448

-----

#### 40. Re: Parallel vs Serial Tone

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>  
Date: Mon, 17 Jul 1995 09:26:58 +0200 (MESZ)  
To: hfsig@tapr.org  
Message-ID: <199507170726.JAA15524@hades>  
Rendered body type: text/plain

Hi all!

Thank you very much for the pleasant discussions during the last weeks. I like to contribute to the discussion of seriell vs. parallel type modems. This discussion is quite popular now in Germany an Europe as we are going to use a parallel type modem (coded OFDM) for broadcasting digital audio (DAB). There were several articles stating that single carrier transmissions are better in general. As our institute here does research work on OFDM this is of course not our opinion. Looking at HF there were a few comparisions of single tone modems to parallel tone ones. All this papers have in common that the modems are compared using no coding at all. This is not fair at all, as single carrier modems need a demodulation scheme that is somehow a form of coding. I think this is due to the fact that parallel tone modems are older than single carrier ones and therefore usually do not use advanced signal processing algorithms or coding. If you look at recent papers about parallel tone modems there is still enough to do. The only real drawback when using parallel tone modems is the limited mean power with peak power limited equipment. If you are using proper FEC schemes, both types should be comparable even at low BER. From the implementations point of view you need a considerably higher complexity for single carrier modems than for parallel ones. If someone is interested in information look for articles from Clark, A.P. or for: Clark, A.P.: Adaptive Detectors for Digital Modems. Pentech Press, London 1989. He discusses several very interesting aspects of single carrier transmission on HF.

So far no one has mentioned the advantages of parallel tone modems other than the relaxed complexity. We have a back channel from the receiving to the transmitting side which we can use for informing the transmitter of the channel conditions. The receiver could i.e. tell the transmitter, that one frequency bin is jammed, so that the transmitter can leave this bin empty. Try this with a single carrier modem...

BTW. Can anybody make me a copy of the Mil-STD things? It is not so easy to get these things in Germany. Or is there maybe a Gopher server like for the CCITT?

Alexander DL8AAU

--  
\*-----+-----\*  
| Alexander F. Kurpiers |  
| Institut f. Uebertragungstechnik | Voice: +49-6151-162369 |  
| u. Elektroakustik | Fax : +49-6151-165545 |  
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de |  
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0eam.#hes.deu.eu |  
\*-----+-----\*

#### 41. Re: [HFSIG:471] Re: ANDVT TACTERM Documentation

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Mon, 17 Jul 1995 10:22:06 +0200 (MET DST)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.950717092450.2226A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Fri, 14 Jul 1995, Johan Forrer wrote:

>  
> In a document that I received from Phil Karn, it describes in a fair amount

- > of detail, how one outfit did their own 16-tone MIL-STD-188 modem. They also
- > had to deal with this same problem of using readily-available SSB rigs.
- > Their solution was to use an audio band 935 - 2585 Hz for the tones.

Yes, this is the 16-tone modem from MIL-STD-188-110A, Appendix A. The interesting thing is that the ANDVT TACTERM modem is based on the 39 tone modem described in Appendix B. They are using the same preamble tones for both Doppler Preamble and Frame Sync preamble. The length of the preambles are not exactly the same though. And the 16 data tones have been extracted from the 39 tone set.

- >
- > I also believe that there is no question that a single tone system at low
- > baud rate will outperform a parallel tone system (in terms of BER for any
- > given s/n). However, in the end it becomes a tradeoff between speed and
- > robustness. The parallel modem will certainly have the advantage for speed.
- > I think one have to be careful when reading papers about these things
- > without normalizing everything to the same units (BER for given S/N at
- > bits/second/hertz). This is confusing.

I agree it sometimes sounds confusing. I have no guarantee that they did the normalization correct in this paper I referred to, but the results of the modem tests was plotted in the same diagram and as they used SNR instead of Eb/No (bit energy to noise-power spectral energy) I supposed they had normalized for bit rate and bandwidth. I would like to invite all of you to have a close look at Bernard Sklar's tutorial "Defining, Designing, and Evaluating Digital Communication Systems", IEEE Communications Magazine, Vol. 31, No. 11, Nov. 1993, pp.92-101. I find this an excellent paper in clarifying a somewhat convoluted subject.

As I have already put on my teaching hat I will give you a few other literature pointers as well. First of all I should like to mention the paper by Joseph M. Perl, "Channel Coding in a Self-Optimizing HF Modem", Proceedings of the 1984 International Zurich Seminar on Digital Communications, March 6-8, 1984, (IEEE Catalog No. 84CH1998-4), pp.101-106. This is an interesting approach for designing an adaptive and optimizing HF modem. It can choose between four different modulation types: DPSK, DQPSK, FSK and MFSK, while the number of tones are equal or less than 49, baud rate equal or less than 100 Baud, in band diversity, error correcting codes (Golay 1/2; Majority, Convolutional 1/2, 1/3, 1/4), interleaving delay equal or less than 4.5s, with hard and soft Viterbi decoder.

The trick is of course the Link Quality Evaluation (LQE) which is based on analysis of the received information or on the preamble tones, hence no special link quality signaling is needed. The channel estimation does not take more than about 8% of DSP processor's resources. The channel evaluation techniques used are described in another paper, which I don't have the reference to where I am sitting at the moment. I can come back with that if there is an interest.

Another paper that almost answers Pawel's questions about optimum symbol rates etc. is: Doron Rainish and Joseph M. Perl, "Generalized Cutoff Rate of Time- and Frequency-Selective Fading Channels", IEEE Trans. on Comm., Vol. 37, No. 5, May 1989, pp.449-467. They look at optimal code rate and symbol element duration for MFSK and MDPSK in various HF channels. They also look at optimal guard time. An interesting thing is the special case for MDPSK over the CCIR HF channel, where the optimal guard times equals the multipath spread and is independent of SNR. The paper is at times rather heavy, but the final plots are very clear.

The question of guard time is an interesting one. Ralph is rather sceptical to the value of using guard time, and the paper above also shows that the real value is somewhat dubious. One thing is sure and that it is a loss of signalling energy and that has to be weighted against what effect it can have on combating multipath spread.

The question about what type of modem which is the optimal for amateur use is very difficult to answer as we have very diverse needs. Some of us only want to have a chat now and then and then a MFSK (a la Piccolo) modem should assure good performance under almost all conditions, while others are interested in moving large amount of data at high speed to tie together packet radio networks etc. For the high speed use both parallel tone and serial tone modems have their place and as Peter Reynolds', KE4BAD, paper "HF Channel Simulator Tests of Clover", QEX, December 1994, pp.7-12 showed a serial tone modem like the one described in STANAG 4285 is a very serious contender. It clearly beats the Clover waveforms.

73's Frode

Frode Weierud                      Phone : 41 22 7674794  
CERN, SL                              Fax : 41 22 7679185  
CH-1211 Geneva 23, Switzerland   E-mail : frode@dxcern.cern.ch

---

## 42. Re: [HFSIG:448] Re: HFSIG activities

From: JALOCHA@chopin.ifj.edu.pl  
Date: Mon, 17 Jul 1995 12:21 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <C84983ED6023255A@chopin.ifj.edu.pl>  
Rendered body type: text/plain

I am very pleased to announce that I have made a useable modem's code with KISS interface.

Just to remind you:

My modem uses 15 tones spaced at 150 Hz. Each tone is DQPSK modulated at 100 baud. Total bit rate is thus  $15 \times 100 \times 2 = 3000$  bps. The modem auto-tunes to incoming packets within at least +/- 75 Hz ( +/- 100 Hz is the absolute limit ). Optionally you can activate simple FEC: for every 11 data bits you add 4 and afterwards you can correct single bit errors. Interleave is the other option: any reasonable factor can be specified.

The modem is still missing frequency and symbol timing tracking during the data decoding phase. My initial tone's phases aren't very good either so I get peak/RMS = 4 (12 dB).

So far I have only made a simple loopback tests and I get the following results:

FEC off, S/N = 13 dB => 3/4 of 230 byte packets get through

FEC on, S/N = 7 dB => 3/4 of 230 byte packets get through

Don't take these results too serious, as the noise spectrum wasn't very flat...

Noise and RMS measurements were done with a soundcard by recording the noise/signal and computing the RMS off-line. Packet loss rate was measured with JNOS by sending out regular beacons and looking up the number of received packets.

There is a hope that the first on-air tests will be done this week by Timo and Joni from Finland.

Pawel

---

## 43. Re: [HFSIG:474] CLOVER II vs. Pactor II

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 17 Jul 1995 09:38:34 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <580223A120B@frl.orst.edu>  
Rendered body type: text/plain

Hi Rick,

You pose some difficult questions. The probably are aware of a lot of claims are presently being made as well as some degree of personalities involved. I do not wish to add to the confusion, however, just add my two cents worth.

I am of the opinion, at least as far as I can tell of this group of technically-minded participants, that all are interested in something that shows potential, however, without some common-sense evaluation or theoretical proof, who really knows what is better?

There was a posting about a year or so ago from Rolf Sommerhalder about several systems that werer compared using an HF channel simulator - these included Pactor I and Clover II. There also was an interesting paper in QEX a while ago, mostly on Clover's capabilities - also done using a channel simulator. It may be worth your while finding these and perhaps it will give you a bit further insight - this the way how modems should be evaluated. From this you would want to see the BER vs. S/N curves.

IMHO, the designers of Pactor I must be credited for clever innovation, basically on what was available at the time, i.e., 100 baud FSK. The protocol does, however, have a few snags. (a) a totally inappropriate

synchronisation method, (b) no automatic link re-establishment after failure, (c) a flawed fall-back to lower rate when conditions are really bad. The claims of the designers regarding memory ARQ gain, was based on AWGN and not quite appropriate. Implementation of memory ARQ by other licencees not using any form of A/D is rather amusing. Pactor II has several innovative features such as stronger ECC and a new modulation. With this innovation comes new challenges - frequency accuracy needs closer tolerances - fallback procedures are getting more complex. Time will tell how practical it all will work out. I can only guess the results when a device like this is being used by a complete non-technical novice.

I don't know how many folks have followed the evolution of Clover from the early days of CCW? Again, my personal opinions - I have always have had a great deal of respect for Clover's principles and believe it to be superior to anything that we have had on the ham bands. That is due to its modulation format, ECC, and also the transmission protocol is well engineered.

Of the two, Pactor II and Clover II, my personal opinion is that the Clover modulation is superior to Pactor II's (I think it may be possible to prove it theoretically by deriving each error function (Marquum Q)). Pactor II, however, probably have a stronger ECC. We will have to see how these two utilities plays off against each other, i.e., Clover's time/frequency diversity principles against Pactor II's convolution code.

I still am asking the unpopular question: why have we not seen a technical disclosure of either Clover or Pactor's such as we have seen done for G-TOR for example (published in full in the 1994 DCC proceedings). I mean at a level sufficient that amateurs can evaluate how things are really working.

Sorry that I cannot offer a clear answer - I am afraid I only add to the confusion.

73's

--Johan

As far as I can see, not many of the participants in this net will jump in to buy, but rather

---

#### 44. Re: [HFSIG:478] Re: Parallel vs Serial Tone Modems

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Mon, 17 Jul 95 17:16:26 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sXySg-0001mUC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

Charles,

In your message of 17 Jul 1995 at 0134 CDT, you write:

>  
> >I do not mean to stifle efforts in developing a serial tone modem...  
> >I'm just trying to be a little practical.  
> >73, Walt  
> >  
> >  
> Hey,  
> Walt we do things because they are there!!! Seriously I have been waiting for  
> a discussion on serial tone modems. A simple slow speed modem appears to  
> be quite do-able. The main problem is I think is the probable requirement  
> to use  
> multiple DSPs.

Rgr Rgr...don't let me slow you down and if I can duplicate  
your effort, rest assured I will.

> I have been reading about the equalisers used in mobile radio modems, I guess  
> the real difference is the actual multipath delays are much greater

> therefore longer  
> equalisers are needed for H.F.  
> Whether we use a training sequence or decision directed I cannot guess.  
> However it may well be that a slow < 1kbit/s modem is more suitable for the ham  
> environment especially something that is proof against those pesky CWers!  
>  
> Something like a 300 baud modem would be interesting with a simple equaliser  
> in front of it.

No doubt about the need for a \*very\* robust modem for those occasions when the noise is high and/or the signal is low and you need to get a message thru, then this type of modem would be nice. Its something we might use here on the Texas Gulf Coast during a hurricane.

>  
> As far as receivers are concerned a nice project would be to take an I.F signal  
> down convert it to say 10Khz then demodulate using DSP. I will have to  
> investigate my TS850 as it does something like that for it's external DSP unit.  
> (Although it just filters).  
>  
> Another thing would be an intelligent VOX with compression done in DSP  
> could event have an expander at the far end.  
>  
> I must admit I have learn't a lot the last few months from this news group, my  
> 8ary modem is proceeding it has been re-written a couple of times due to  
> my improving knowledge. My main problem currently is gaining and tracking  
> bit sync. The sync has to lock as fast as possible. It should also have the  
> ability to re-sync to a colliding signal this of course causes a problem!  
> The technique I am using is to divide the tone with the highest power with  
> that of the others. This produces a nice triangle wave. I then use an early/late  
> gate to find the peak. This locks the NCO (thats the bit I can't get to work  
> yet)  
> the NCO then triggers an integrate and dump. At the end of the period a decision  
> is made as to which tone was received.

>  
>  
> Regards Charles

> -----  
> "projects don't slip, they just catch up with reality"  
>  
> Charles Brain (G4GU0)  
> Chelmsford, Essex.  
> E-mail chbrain@dircon.co.uk  
> POTS +44 (0)1245 353221  
> FAX +44 (0)1245 275448  
> -----  
>

Keep up the good work and say HOWDY, to all tha hams at the Chelmsford Radio club and the Marconi folks. I really miss G0AEH, Albert, from Blackmoore/Hook-End...he was a very nice fellow.

## 45. Re: [HFSIG:482] Re: CLOVER II vs. Pactor II

From: muphaus@cris.com (Marv Uphaus)  
Date: Mon, 17 Jul 1995 17:16:24 -0500  
To: hfsig@tapr.org  
Message-ID: <4EuCwM82cese083yn@cris.com>  
Rendered body type: text/plain

Johan...

On Mon, 17 Jul 1995 11:44:13 -0500, you wrote:

>I am of the opinion, at least as far as I can tell of this group  
>of technically-minded participants, that all are interested in something  
>that shows potential, however, without some common-sense evaluation  
>or theoretical proof, who really knows what is better?

I just want to add some thoughts of my own...

A commonly held opinion of hams is "What's my discount"... Hams always want the cheapest, bestest thing they can get... Several things come into play here... Cheap, if it works, will always win out over expensive... This has been, IMHO, the reason Clover has not taken off... This new Clover board may be the breakthrough that Clover needs... But \$395 is still a lot... Look at the DigiCom packet stuff for the Commodore... That stuff is still running around the world... Good and cheap... Look at the success of BayCom... Look at the success of Pactor I... I tuned around

recently on a weekend on 20 meters and Pactor I is the only thing that I heard... Why was it successful... It was pretty cheap and everyone had it... Once AEA and MFJ and Kantronics all put it in their products it became a standard... Until somebody besides HAL sells Clover it will never gain popularity, even if it is 1000 times better... G-TOR has a good chance since it is being licensed...

None of this stuff is going to ever be popular on Ham Radio unless someone opens up the licensing... We would probably not have packet radio now if it hadn't been for TAPR... All the current popular modes on ham radio are open technically and copyright wise... The great success of the IBM PC had largely to do with the open architecture of the buss... The only good thing IBM ever did for the computer community, in my mind... (Don't flame, I won't read or respond...)

>I still am asking the unpopular question: why have we not seen a  
>technical disclosure of either Clover or Pactor's such as we have seen done  
>for G-TOR for example (published in full in the 1994 DCC proceedings). I  
>mean at a level sufficient that amateurs can evaluate how things are  
>really working.

One of the things that I hope will come out of this group is a definitive statement concerning a good CHEAP way for amateurs to communicate digitally via HF... I think that we are on the way... I hope when something is decided that TAPR might pick up the ball again... I think that using the sound cards is a BIG step forward... This group ought to be the ones to put together the digital basis for HF communications into the 21st century... Something that any moderately technical ham can get running for less than \$200...

I have an old HAL RTTY modem, a CRI-200, I found at a hamfest for \$45... That and BMK Multy got me on RTTY, PACTOR, AMTOR for less than the \$200... I would do it again... We need this kind of thing for advanced HF digital and I am convinced that it is do-able... Johan is there with the PC-TOR stuff, but it needs to be with the more robust error correcting and higher data speeds... Lets keep pushing forward...!!!

In the mean time Internet Relay Chat and E-Mail on the Internet is taking over the role of digital communications via ham radio... And on IRC you don't have to know the code or take an FCC test... There are thousands of technically competent folks using that now, who could be hams... There's even a ham radio area (#hamradio) on IRC, populated by a BOT and people...

We need to move forward or ham radio is going to flounder and forever be gone...!!! I'm on the Internet every day and on ham radio very rarely... I don't want to see it go, but I think that it's demise is getting closer...

73... Marv... K4BVG

In the midst of great joy, do not promise anyone anything. In the midst of great anger, do not answer anyone's letter. - Chinese Proverb

-- Marv Uphaus -- muphaus@cris.com -- finger for PGP public key --

---

## 46. Re: Parallel vs Serial Tone

From: esilbaug@hawkeye (Eric Silbaugh)  
Date: Wed, 19 Jul 95 10:39:59 EDT  
To: hfsig@tapr.org  
Cc: esilbaug@afit.af.mil  
Message-ID: <9507191439.AA03120@hawkeye>  
Rendered body type: text/plain

Here is a reference for those of you interested in comparing the COFDM and single-tone techniques which Alexander, DL8AAU, discussed in his recent posting.

H. Sari, G. Karam, I. Jeanclaude, "Transmission techniques for digital terrestrial TV broadcasting", IEEE Communications Magazine, February 1995, pp.100-109.

The authors begin with an overview of OFDM and channel equalization. It turns out that OFDM is very similar to a single-tone system using frequency-domain equalization. This was interesting since most equalization techniques I have seen (including those used in the single-tone MIL-STD modems) work in the time-domain. In retrospect I guess it shouldn't be surprising that equalization can be done in the frequency-domain considering all the time-frequency dualities in Fourier analysis.





>2\*(500+300)=1.6 KHz of bandwidth. Is this right?

Correct - that is roughly how I interpret part 97 as well. One further point that is often forgotten, is that part 97 also specifies which code alphabet (i.e. ASCII or BAUDOT). The latter has clearly been violated by modern digital modes - Clover for instance by using ECC, Pactor and G-TOR when using data compression - just a technicality that have never legally been challenged. This, however, is if you interpret the law to the letter. Clearly, however, the mentioned violations are not serious, and as a matter of fact not intentional encryption. One might argue that it is done all the time - that is not how part 97 is written - I suspect that the matter about the code alphabets will rest unless challenged. IMHO, it will be a total waste of time and resources if it does become a legal issue.

I would rather see that our FCC regulations regarding the digital modes be brought up to date with the UK and German regulations, or otherwise to keep track with modern technology.

>Next Question. If 1.6 KHz is right, why does Clover fit into 500 Hz? Are  
>we required to use CW type bandwidths? If not, then why bother with such  
>tight restrictions since wider bandwidth is (probably) easier to use  
>effectively...except for the CW interference problems.  
>Thanks  
>Rick W6NZK  
>-----  
>Rick Booth  
>E-mail: rick@itron-ca.com  
>-----  
>  
>

That is a real issue.

Yes, Clover technically fits into a 500 Hz slot - Pactor II's waveform is also shaped to fit in a 500 Hz slot. This pretty much follows a belief that narrow-band emissions conserve spectral space.

With the new high speed modems coming along, something inevitably will have to happen. I hope that this evolution happens in a way that will have the least amount of negative impact. I do suspect, however, that the notion will be fought every inch of the way. In short, yes, we probably can squeeze a number of orthogonal channels into 1 KHz and comply to the present regulations. It would be better than anything we have at the moment, but not a good long term solution.

It should be our mission to convince the digital community to support our efforts. I don't think that many know what we are intent on putting on the air, i.e., network vs keyboarding, modulation format and it's occupied bandwidth. The concepts must be formalized in clear language.

Without spending to much time phrasing this correctly - this is what I had thought about (of course without saying - open for debate, my own personal opinions):

First, we have to show that we have superiour technology, i.e., it gets traffic through, even under adverse conditions without making dreaded demands on spectral usage. Instead of beating 300 baud packet to death, we need to illustrate an effective alternate solution.

Second, we have to convincingly show that narrow-band emissions are as prone (probably more) to causing interference and suffering from interference than does our proposed wide-band emissions. This holds for man-made, or HF propagation wise.

For an interesting aside: the argument that the famous John Costas had to defend when he was making attempts to convince the IEEE of why wide-band emissions, was quite similar - this was when spread-spectrum communications was born. I am certain that I don't have the vision and influence that Costas had, but it is only a matter of time. You already know the outcome of that one.

To help the argument along, just for starters consider the following: The impact on other nearby stations for a digital modulated 100W packed into 500 Hz (0.2W/Hz) compared to 100W spread over 3KHz (0.033W/Hz). Even with the best filters,

how wide will a high-powered signal really be? The main concern that I have regarding wider bandwidth emissions is when an operator indiscriminantly puts a big linear behind such a signal - this would be disasterous and counterproductive.

The notion of low power/Hz, and ideally lots of Hz, is what it's all about. The wider it gets, the better it gets (give and take a bit of liberty). Our present parallel modems does not use a spreading function as in SS yet - at the very highest rate it needs all the bandwidth, but at lower rate frequency diversity is used, in which case bandwidth is used with redundancy.

Another example: Would the principle ideas as used in the cellular phone market have worked if it was based on narrow-band transmission?

What is needed is an effective demonstration of what we are intending to use and win the support of the digital community.

-----  
The real issue and crux of the matter is how to best utilize our allocated spectrum by a number of simultaneous users without QRM (leave poor operator judgement aside for the moment). Weigh the advantage of replacing the current HF digital networks with an efficient shared-channel mode.  
-----

I expect that this issue will stir up a lot of discussion. This is an invitation to hear your views on how this should be approached.

73's

--Johan

---

#### 48. Re: [HFSIG:486] FCC Regulations for HF digital

From: Greg Jones <wd5ivd>  
Date: Wed, 19 Jul 1995 21:20:54 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199507200221.VAA10073@dingus.n5lyt.datarace.com>  
Rendered body type: text/plain

Hi Johan -- has anyone checked the rules very recently. In a thread on netsig we had this posted.

```
>> EVERYBODY is using AX.25 because the FCC says you MUST.
>
>The AX.25 requirement has been gone for about a year, and I didn't hear
>about it when it happened either.
>
>This is the new 97.109e:
>
> (e) No station may be automatically controlled while transmitting
> third-party communications, except a station participating as a
> forwarding station in a message forwarding system.
>
>This paragraph used to contain the AX.25 requirement. It's gone.
>
> Bruce Perens
```

While this does not directly impact HF modes, it does impact networking and BBS operations.

The main point is that the FCC, when aware of something needing to be done, will change the rules when a section is open.

So - we should think about talking to the TAPR FCC Reg committee and some of the people at the league about this issue so that they can be aware of it. I would think that Paul Rinaldo is aware of it -- the reason that many of the alternate protocols/modes were published in the last Digital Communications Conference Proceedings was to be able to have a reference to then include under some 'misc' section of the rules.

I forget all the maneuvers required. Phil is on the Future Systems Committee and might be able to comment....??

> First, we have to show that we have superiour technology, i.e.,  
 > it gets traffic through, even under adverse conditions without  
 > making dreaded demands on spectral usage. Instead of beating 300  
 > baud packet to death, we need to illustrate an effective  
 > alternate solution.  
 >  
 > Second, we have to convincingly show that narrow-band emissions are  
 > as prone (probably more) to causing interference and suffering from  
 > interference than does our proposed wide-band emissions. This holds  
 > for man-made, or HF propagation wise.  
 >  
 >  
 > -----  
 > The real issue and crux of the matter is how to best utilize  
 > our allocated spectrum by a number of simultaneous users without  
 > QRM (leave poor operator judgement aside for the moment). Weigh  
 > the advantage of replacing the current HF digital networks with an  
 > efficient shared-channel mode.  
 > -----  
 >  
 >  
 > I expect that this issue will stir up a lot of discussion. This is an  
 > invitation to hear your views on how this should be approached.  
 >

This will indeed be an interesting debate in the future. It already has  
 been.

Personally, I think we continue to improve the technology. If we can make  
 HF communicatins work at better rates with better reliability I think we MUST  
 advance the state of the amateur art.

The problem is that you can spend lots of time worrying about others. While  
 all efforts should be made to communicate the technical advances -- for  
 every amateur willing to listen and understand there will be two more  
 sending flames. However unfortunate -- a common amateur trait. All of  
 amateur history is filled with examples: ssb/am, etc.

The golden rule of amateur radio -- "those that develop hardware or software  
 'rule'" Pretty simple. If you develop something that works better, faster,  
 and hopefully (for most hams) cheaper :-) then no matter how much yelling,  
 people will start to use it. Will it impact others -- yes. Like any mode in  
 amateur radio.

However, limitation of modes or speeds is not the correct manner in  
 which to enforce better operating correctness.

The problem is that it is easier to regulate then educate. Amateurs might  
 be good at communications technology, but are in general terms poor  
 communicators.

Well, I'll get off my soap box.

I hope the HF SIG keeps up the advancement of the amatuer art. The rules  
 will eventually sort them self out. We can work with the FCC to make sure  
 that what we are doing is seen as within the scope of the rules.

I must congratulate everyone on this SIG. I feel that this is one of the  
 best SIGs we have in TAPR. The level of discussion and technical research  
 is terrific to see happening! Hats off to Johan and everyone else.

Cheers - Greg, WD5IVD

---

## 49. FCC part 97 on WWW?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
 Date: Thu, 20 Jul 1995 10:24:00 PST8PDT  
 To: hfsig@tapr.org  
 Message-ID: <5C8E62F48E5@frl.orst.edu>  
 Rendered body type: text/plain

Hi,

Does anyone know whether the FCC have a WEB page and whether  
 perhaps the latest reg's are available in electronic form?

Thanks in advance,

--Johan, KC7WW

---

## 50. Re: [HFSIG:485] Re: Parallel vs Serial Tone

From: mcdermot@eagle.aud.alcatel.com (Tom Mcdermott)  
Date: Thu, 20 Jul 95 15:39:30 CDT  
To: hfsig@tapr.org  
Message-ID: <9507202039.AA01771@eagle.aud.alcatel.com>  
Rendered body type: text/plain

Eric Silbaugh brings up several good points in the debate on multi-carrier vs. single carrier modulation (will this become the SSB vs. AM war of the 1990's ?)

> In the next section the authors present the results of some computer  
> simulations showing that losses produced by frequency offsets and  
> power backoff (due to the high peak amplitudes) in OFDM produce BERs  
> much larger than those of an equalized single-tone system.  
>

It should be noted that high orders of modulation on single carrier transmission (such as 64 QAM, etc.) also require a great deal of linearity in the transmitter. It can become debateable as to which modulation format imposes the more severe linearity requirement!

> One observation the authors make is that frequency selective fading  
> produces an irreducible BER in OFDM systems not using forward error  
> correction (FEC). Which leads to their last results showing that an  
> OFDM system with FEC, coded OFDM (COFDM), can produce BERs identical  
> to or slightly better than an equalized single-tone system (without  
> FEC). At the very least this shows the power, and necessity, of FEC!  
>

With HF, time-diversity is an easily achievable diversity (and with HF, any kind of diversity helps greatly!). Recent work on single carrier modems has focused on equalization using minimization of the Viterbi error metric, rather than minimization of the ISI, as a good variable to reduce when adjusting the adaptive equalizer. This of course presumes convolutionally coded data. Not sure which modulation format requires the greater computational load for equalization. Custom silicon does this very well in HDSL and ADSL land-line systems, and the equalizers are big.

> One unanswered question is the relative implementation complexity  
> (DSP cycles) of COFDM versus single-tone modulation with frequency-  
> domain equalization. All the time domain equalizers I have seen are  
> tremendous cycle hogs (not the Harley-Davidson kind either). I  
> wonder if frequency-domain equalization is any easier. Can a DSP  
> sound card handle it?  
>

> It also seems that performance of the single-tone modem could be  
> improved by adding FEC. Of course, this would require even more DSP  
> cycles, unless the host computer does the FEC decoding.  
>

> COFDM would seem to have the initial advantage in reduced complexity  
> (please correct me if this is wrong). Past examples of working  
> systems show that it can be done. As long as the carrier tracking  
> and high peak power limitations can be overcome it looks like this  
> could produce a robust system.  
>

> As others have noted a fairly inexpensive, robust, moderate speed  
> HF modem will be a big step forward. Basing it on DSP just means it  
> can be improved without buying new hardware. Keep up the good work!  
>

> Eric, N2NNP

> . . . . .  
> : ' : ' : Eric E. Silbaugh  
> : . . . : esilbaug@afit.af.mil  
> : : :  
> : : : All standard, non-standard, and  
> : : : MIL-STD disclaimers apply  
>

-----+-----  
Tom McDermott  
Alcatel Network Systems, Inc.  
mcdermot@aud.alcatel.com

| "All opinions expressed  
| are my own, and do not  
| represent those of Alcatel

## 51. Re: [HFSIG:488] FCC part 97 on WWW?

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Thu, 20 Jul 95 17:34:00 -0500  
To: hfsig@tapr.org  
Message-ID: <m0sZ4AL-0001zcC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

Johan,

In your message of 20 Jul 1995 at 1239 CDT, you write:

> Hi,  
>  
> Does anyone know whether the FCC have a WEB page and whether  
> perhaps the latest reg's are available in electronic form?  
>  
> Thanks in advance,  
>  
> --Johan, KC7WW  
>

For the FCC try:

[http://fcc.gov:70/0h/AAA\\_HOMEPAGE.html](http://fcc.gov:70/0h/AAA_HOMEPAGE.html) or <http://www.fcc.gov>

For Part 97 (the ARRL Official Copy) Try:

<http://www.acs.oakland.edu/barc/arrl/part97.html>

or <ftp://arrl.org> and I think you can ftp down part 97.

I have visited [www.acs.oakland.edu](http://www.acs.oakland.edu).

Walt

## 52. Fed Std 1052

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Tue, 25 Jul 1995 19:02:00 +0100  
To: hfsig@tapr.org  
Message-ID: <199507251954.AA26597@felix.dircon.co.uk>  
Rendered body type: text/plain

Hi,

Does anyone know the current standing of Fed-Std 1052 Appendix B  
I think (The data link protocol). I have a 1992 version of it I was just  
wondering  
if it has moved from proposed to an actual standard. I am also interested in  
Fed-Std 1047, it was mentioned in QEX a few months back, I would like to read  
it, need something to help me sleep at night.  
There must be a way of finding out which standards are current.

Another thing anyone managed to get IIR filters to work on a TMS320C50 all  
mine oscillate. I have been using PC-DSP to generate the coefficients and the  
code in the TI C50 databook, all I get get is wonderful oscillation!

Regards Charles.

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail [chbrain@dircon.co.uk](mailto:chbrain@dircon.co.uk)  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

## 53. Bandwidth of digital modulation

From: Alan Bloom <[alanb@tsnake2.sr.hp.com](mailto:alanb@tsnake2.sr.hp.com)>  
Date: Tue, 25 Jul 1995 13:09:14 -0800 (PDT)  
To: hfsig@tapr.org  
Message-ID: <199507252009.AA056022955@tsnake2.sr.hp.com>  
Rendered body type: text/plain

I ftp'd an up-to-date (April 26, 1995) copy of part 97 from [oak.oakland.edu](http://oak.oakland.edu).

It's well hidden: /pub/hamradio/arrl/infoserver/rib/part97.txt

The bottom line is that there are no explicit bandwidth limitations below 28 MHz. There is a maximum FSK shift of 1 kHz and a maximum symbol rate = of 300 baud -- I suspect that the FCC expected that would impose a = de facto bandwidth limitation based on 97.307(a), but they were = probably not thinking about multi-carrier modulation!

AL N1AL

-----  
-

Subpart D--Technical Standards  
=2E...

S 97.305 Authorized emission types.  
=2E...

[Lists legal modulation modes for each frequency sub-band, =  
with footnotes referencing paragraphs in section 97.307(f)]

S 97.307 Emission standards.

(a) No amateur station transmission shall occupy more bandwidth than =  
necessary for the information rate and emission type being transmitted, i=  
n =

accordance with good amateur practice.  
=2E...

(f) The following standards and limitations apply to transmissions on =  
the frequencies specified in S 97.305(c) of this Part.

[Phone bands below 29.0 MHz:]

(1) No angle-modulated emission may have a modulation index greater =  
than 1 at the highest modulation frequency.

[Phone bands below 225 MHz:]

(2) No non-phone emission shall exceed the bandwidth of a =  
communications quality phone emission of the same modulation type. The =  
total bandwidth of an independent sideband emission (having B as the firs=  
t =  
symbol), or a multiplexed image and phone emission, shall not exceed that=  
=  
of a communications quality A3E emission.

[CW bands below 28 MHz:]

(3) Only a RTTY or data emission using a specified digital code liste=  
d =  
in S 97.309(a) of this Part may be transmitted. The symbol rate must not=  
=  
exceed 300 bauds, or for frequency-shift keying, the frequency shift =  
between mark and space must not exceed 1 kHz.

[28-28.3 MHz:]

(4) Only a RTTY or data emission using a specified digital code liste=  
d =  
in S 97.309(a) of this Part may be transmitted. The symbol rate must not=  
=  
exceed 1200 bauds. For frequency-shift keying, the frequency shift betwee=  
n =  
mark and space must not exceed 1 kHz.

[50.1-54 MHz, 144.1-148 MHz:]

(5) A RTTY, data or multiplexed emission using a specified digital code listed in S 97.309(a) of this Part may be transmitted. The symbol rate must not exceed 19.6 kilobauds. A RTTY, data or multiplexed emission using an unspecified digital code under the limitations listed in S 97.309(b) of this Part also may be transmitted. The authorized bandwidth is 20 kHz.

[222-225 MHz, 420-450 MHz:]

(6) A RTTY, data or multiplexed emission using a specified digital code listed in S 97.309(a) of this Part may be transmitted. The symbol rate must not exceed 56 kilobauds. A RTTY, data or multiplexed emission using an unspecified digital code under the limitations listed in S 97.309(b) of this Part also may be transmitted. The authorized bandwidth is 100 kHz.

[Amateur bands above 902 MHz:]

(7) A RTTY, data or multiplexed emission using a specified digital code listed in S 97.309(a) of this Part or an unspecified digital code under the limitations listed in S 97.309(b) of this Part may be transmitted.

[51-54 MHz, 144.1-148 MHz, >222 MHz:]

(8) A RTTY or data emission having designators with A, B, C, D, E, F, G, H, J or R as the first symbol; 1, 2, 7 or 9 as the second symbol; and D

or W as the third symbol is also authorized.  
=2E...

[Paragraphs 9, 10, and 11 deal with novice CW privileges and phone operation in regions 1 and 3.]

[>902 MHz:]

(12) Emission F8E may be transmitted.

[219-220 MHz:] =

(13) A data emission using an unspecified digital code under the limitations listed in S 97.309(b) of this Part also may be transmitted. The authorized bandwidth is 100 kHz. =

S 97.309 RTTY and data emission codes.

(a) [Specifies legal codes: 5-unit Baudot, 7-unit AMTOR, 7-unit ASCII]

(b) Where authorized by S 97.305(c) and 97.307(f) of this Part, a station may transmit a RTTY or data emission using an unspecified digital code, except to a station in a country with which the United States does not have an agreement permitting the code to be used. RTTY and data emissions using unspecified digital codes must not be transmitted for the

purpose of obscuring the meaning of any communication. ...

[end]

---

## 54. Re: [HFSIG:491] Fed Std 1052

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 25 Jul 1995 13:48:24 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <64451CC67C0@frl.orst.edu>  
Rendered body type: text/plain

Hi Charles,

Re: IIR filters. If you have a pole(s) close to the unity circle that may happen. Also if you are aiming for very high Q's, i.e., narrow passbands with brickwall filters, this may lead to that situation. For IIR filter architectures, part of the filter's output is determined both by past input as well as past output. It is not difficult to see that it doesn't take much to get the output history overpower the output history. Typically, once oscillation starts, you may even take the input to zero and it will merrily feed back output history and maintain the oscillatory state. The dampening factor is supposed to take care of this.

Just a thought: do you clear the input and output history on initialization, or do you just leave use the garbage that is in memory? Does it start up OK and starts oscillation only when it sees a large signal? In the W9GR software, you can quite easily get the IIR input filter to saturate by driving the audio input to hard, however, there is sufficient dampening to recover - one can see it return to normal, however like slow release AGC.

Numerical roundoff error is another issue that may add to your problem: since you only have a limited wordsize to work with, it is inevitable that you will encounter roundoff errors for each result that is computed. Right? The bad news is that each roundoff error are fed back and used as part of the input to the filter on the next round, so you have these errors in a recursive positive feedback loop! Certainly not an additive noise source like for instance FIR filters.

There are several IIR filter designs around. Those that use an analog counterpart, does pre-warping, and then the transformation. This is a common way of doing them and I suspect that is what you may be using. There are some other exotic means of designing IIR filters, though I have never attempted that.

I don't know whether this helped. Good luck anyway.

--Johan, KC7WW

---

## 55. Re:IIR filters an 8ary FSK modem

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Wed, 26 Jul 1995 04:47:35 +0100  
To: hfsig@tapr.org  
Message-ID: <199507260539.AA13589@felix.dircon.co.uk>  
Rendered body type: text/plain

>Hi Charles,

>I don't know whether this helped. Good luck anyway.

>

>--Johan, KC7WW

>

>

Hello,

Well after I twigged that the coefficients are different in the TI description to the

PC-DSP program i.e a coeffs are really the B ones etc, things started to work better. At start up there is no oscillation and whne a signal is in the filters passband there is no problem, however outside the passband the output appears almost totally random and is larger than the wanted signal when tuned to the passband.

I am still looking at the 8 ary modem, it seems to work with a 32 point FFT fed with real values. Using FIR filters pushes the available processing, so I thought I would use IIR as that is what I started with and got nowhere!

It would be nice to get something that would work on the DSP93, the FFT approach is fine on a C50 but I have my doubts about the C25 as I don't think there would be enough cycles.



I know Fredricks managed to get it working with just a C12 so it is possible.

I was wondering what CAD programs people are using, as I mentioned I am using PC-DSP and I have heard MATLAB being mentioned, just curious.

Regards Charles

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 56. Re: [HFSIG:494] Re:IIR filters an 8ary FSK modem

From: mcdermot@eagle.aud.alcatel.com (Tom McDermott)  
Date: Thu, 27 Jul 95 08:05:49 CDT  
To: hfsig@tapr.org  
Message-ID: <9507271305.AA01613@eagle.aud.alcatel.com>  
Rendered body type: text/plain

> Hello,  
> Well after I twigged that the coefficients are different in the TI  
> description to the  
> PC-DSP program i,e a coeffs are really the B ones etc, things started to  
> work better. At start up there is no oscillation and whne a signal is in the  
> filters passband there is no problem, however outside the passband the  
output  
> appears almost totally random and is larger than the wanted signal when  
tuned  
> to the passband.  
> I am still looking at the 8 ary modem, it seems to work with a 32 point FFT  
> fed with real values. Using FIR filters pushes the available processing, so  
I  
> thought I would use IIR as that is what I started with and got nowhere!  
> It would be nice to get something that would work on the DSP93, the FFT  
> approach is fine on a C50 but I have my doubts about the C25 as I don't  
think  
> there would be enough cycles.  
> I know Fredricks managed to get it working with just a C12 so it is  
possible.  
>  
> I was wondering what CAD programs people are using, as I mentioned I am  
> using PC-DSP and I have heard MATLAB being mentioned, just curious.  
>  
> Regards Charles

Charles: a couple of points you might look for:

IIR filters, as Johan mentions, are sensitive to numerical errors due to the feedback. You should make sure that you are rounding, not truncating during your calculations (there's some flags in the TMS320 to control this). Secondly, all intermediate calculations should be done with 32-bit arithmetic, not 16 bit arithmetic, and the intermediate results must be stored as 32-bit numbers. Additionally, you should set the TMS320 flags so that the accumulator clamps at the rail, rather than rolling-over.

And, as Johan has mentioned, if there are any poles close to the unit-Z circle, then the required processing accuracy can even exceed 32 bits.

You should be very careful in your quantization of the coefficients. I would recommend using 32-bit coefficients in many cases. It may not be acceptable to take whatever you get from your CAD program and truncate the coefficients to 16 bits. Your CAD program may or may not calculate the coefficients with adequate precision for the application.

FIR filters, of course do not in general require such high precision, but IIR filters do take work to 'tame'. Good luck.

- Tom, N5EG

-----+-----  
Tom McDermott

| "All opinions expressed

|                                         |                            |
|-----------------------------------------|----------------------------|
| Alcatel Network Systems, Inc.           | are my own, and do not     |
| mcdermot@aud.alcatel.com                | represent those of Alcatel |
| [ ICC '96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]       |                            |

-----+-----

## 57. Re: [HFSIG:492] Bandwidth of digital modulation

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Thu, 27 Jul 1995 08:55:41 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <66F71A05C4B@frl.orst.edu>  
Rendered body type: text/plain

Hi Al,

-----some lines deleted-----

>I ftp'd an up-to-date (April 26, 1995) copy of part 97 from  
oak.oakland.edu.  
>It's well hidden: /pub/hamradio/arrrl/infoserver/rib/part97.txt  
>  
>The bottom line is that there are no explicit bandwidth limitations below  
>28 MHz. There is a maximum FSK shift of 1 kHz and a maximum symbol rate  
>of 300 baud -- I suspect that the FCC expected that would impose a  
>de facto bandwidth limitation based on 97.307(a), but they were  
>probably not thinking about multi-carrier modulation!  
>  
>  
>AL N1AL  
>

----further lines deleted---

You make a good point. To elaborate a little further on this  
issue, here is how HAL justifies Clover II. I quote literally from an  
article from Communications Quarterly by Bill Henry, K9GWT, and Ray Petit,  
W7GHH.

"For those who may be wondering how  
this complex modulation fits Part 97 of the  
FCC Rules and Regulations, let us assure  
you that all CLOVER modes are indeed in  
conformance with existing rules. CLOVER  
passes only one data stream and is therefore  
not a "multiplex" modulation format. The  
CLOVER modulation output is audio tones  
that are used to drive the output to an SSB  
transmitter. This is CCIR mode "J2,"  
which is allowed. The CLOVER base data  
rate is 31.25 bits-per-second -- well within  
the 300 baud maximum HF limitation. The  
CCIR emission designation for CLOVER-II  
is "500HJ2DEN".

OK - this obviously deals with a number of issues that we also have been  
discussing. Comments?

I wish to prepare a few things for the upcoming Digital Conference -  
one is to initiate the motion for updating the rules on digital  
communications. I sense that there is willingness to accept the fact that  
technology has advanced. Please think a bit about what the real issues as far  
as rulemaking are concerned to reflect best the needs for modern digital  
HF communications. I will bring that up with the appropriate person(s). Keep  
in mind that we are living in a world community, so I also would like to  
hear what the regulations allow in other countries in this regard.

Regards,

--Johan, KC7WW

## 58. Help !!!

From: Siddhartha Bhattacharjee <sid@doe.ernet.in>  
Date: Fri, 28 Jul 1995 09:50:17 +0530 (GMT+05:30)  
To: hfsig@tapr.org  
Message-ID: <Pine.3.89.9507280937.B3640-0100000@mahavir>  
Rendered body type: text/plain

Dear friends,

I have got badly stuck up for want of a silly information and hence decided to trouble the HFSIG guys for a possible help.

I need the chip details of XR2206 chip with specific application to the AFSK generation for 200/170 Hz and 1KHz shift for use in HF and VHF use. I tried all possible sources in Delhi and with my friends and colleagues. It appears that the EXAR chips are not very popular here and hence no manual or application note can be found here. I would appreciate an electronic copy of the schematics or the data sheet etc., first uuencoded and then mailed to sid@doe.ernet.in.

Thanks to everybody in anticipation.

73s all round,

Sid.

---

## 59. Re: [HFSIG:465] Re: ANDVT TACTERM Documentation

From: Phil Karn <karn@unix.ka9q.ampr.org>  
Date: Fri, 28 Jul 1995 00:09:48 -0700  
To: forrerj@ucs.orst.edu  
Cc: hfsig@tapr.org  
Message-ID: <199507280709.AAA26888@unix.ka9q.ampr.org>  
Rendered body type: text/plain

>instances. If a "dead" zone or guard interval is used (i.e. the  
>integrator/FFT excuding that part of the signal), he shows a significant  
>improvement in demodulator S/N is possible due to using the guard band.

Seems to me that applying a window function to the FFT in the demod should accomplish this very nicely.

By the way, I see that the bogus "Reply-To: hfsig@tapr.org" header is still present, requiring me to manually edit headers when I reply to send a copy to the person in the From: line...

Phil

---

## 60. Re: [HFSIG:471] Re: ANDVT TACTERM Documentation

From: Phil Karn <karn@unix.ka9q.ampr.org>  
Date: Fri, 28 Jul 1995 00:29:07 -0700  
To: forrerj@ucs.orst.edu  
Cc: hfsig@tapr.org  
Message-ID: <199507280729.AAA26950@unix.ka9q.ampr.org>  
Rendered body type: text/plain

>If the sync pre-amble is designed cleverly, it may be possible to write an  
>algorithm that is hunting for it all the time - a daemon. When it sees it,  
>it forces a frame reset. As an interesting aside, in Pactor, the designers

HDLC flags work exactly like this -- get a flag, go back to the known starting state (and finish whatever packet you were working on). Unfortunately, HDLC flags are much too small to be very robust over radio. A PN sequence seems more reasonable.

Phil

---

## 61. Re: [HFSIG:496] Re: Bandwidth of digital modulation

From: Danny Higgins <Danny@edstks1.demon.co.uk>  
Date: Fri, 28 Jul 1995 08:56:54 GMT  
To: hfsig@tapr.org  
Message-ID: <219@edstks1.demon.co.uk>  
Rendered body type: text/plain

In message <66F71A05C4B@frl.orst.edu> hfsig@tapr.org writes:

> Hi Al,  
>  
> ----some lines deleted----  
>  
>  
>  
> >I ftp'd an up-to-date (April 26, 1995) copy of part 97 from  
> oak.oakland.edu.  
> >It's well hidden: /pub/hamradio/arrrl/infoserver/rib/part97.txt  
> >  
> >The bottom line is that there are no explicit bandwidth limitations below  
> >28 MHz. There is a maximum FSK shift of 1 kHz and a maximum symbol rate

> of 300 baud -- I suspect that the FCC expected that would impose a  
> de facto bandwidth limitation based on 97.307(a), but they were  
> probably not thinking about multi-carrier modulation!  
>  
>  
> AL N1AL  
>  
>  
> ----further lines deleted----

> You make a good point. To elaborate a little further on this  
> issue, here is how HAL justifies Clover II. I quote literally from an  
> article from Communications Quarterly by Bill Henry, K9GWT, and Ray Petit,  
> W7GHH.

> "For those who may be wondering how  
> this complex modulation fits Part 97 of the  
> FCC Rules and Regulations, let us assure  
> you that all CLOVER modes are indeed in  
> conformance with existing rules. CLOVER  
> passes only one data stream and is therefore  
> not a "multiplex" modulation format. The  
> CLOVER modulation output is audio tones  
> that are used to drive the output to an SSB  
> transmitter. This is CCIR mode "J2,"  
> which is allowed. The CLOVER base data  
> rate is 31.25 bits-per-second -- well within  
> the 300 baud maximum HF limitation. The  
> CCIR emission designation for CLOVER-II  
> is "500HJ2DEN".

> OK - this obviously deals with a number of issues that we also have been  
> discussing. Comments?

> I wish to prepare a few things for the upcoming Digital Conference -  
> one is to initiate the motion for updating the rules on digital  
> communications. I sense that there is willingness to accept the fact that  
> technology has advanced. Please think a bit about what the real issues as far  
> as rulemaking are concerned to reflect best the needs for modern digital  
> HF communications. I will bring that up with the appropriate person(s). Keep  
> in mind that we are living in a world community, so I also would like to  
> hear what the regulations allow in other countries in this regard.

>  
>  
> Regards,  
>  
> --Johan, KC7WW  
>  
>

Hi, Johan. You seem to have sent this to me by mistake.

I have found the HFSIG and have about 1 MByte of back discussion to catch up  
with. How do I join in the discussion?

--  
Danny Higgins G3XVR

## 62. Re: [HFSIG:498] Re: ANDVT TACTERM Documentation

From: bm@lynx.ve3jf.ampr.org (Barry McLarnon VE3JF)  
Date: Fri, 28 Jul 1995 17:04:34 +0000 (GMT)  
To: hfsig@tapr.org  
Message-ID: <m0sbsp-0002ruC@lynx.ve3jf.ampr.org>  
Rendered body type: text/plain

> By the way, I see that the bogus "Reply-To: hfsig@tapr.org" header is  
> still present, requiring me to manually edit headers when I reply to  
> send a copy to the person in the From: line...

It's not bogus, it's intentional. The mailing lists that I administer  
are set up the same way, so that the default is for replies to be seen  
by all, and all can learn from them. The underlying assumption is that  
the list is intended for open discussions, and that the majority of  
replies contain information which is of interest to many on the list.  
The downside is the risk of embarrassment when replies which were  
intended to be personal get sent to the list. Still, I think it's a  
reasonable tradeoff on a high-SNR list such as this one.

> Phil

Barry

--

Barry McLarnon VE3JF/VA3TCP  
Ottawa Amateur Radio Club Packet Working Group  
Email: bm@hydra.carleton.ca or bm@lynx.ve3jf.ampr.org

---

### 63. Re: [HFSIG:486] FCC Regulations for HF digital

From: Phil Karn <karn@unix.ka9q.ampr.org>  
Date: Fri, 28 Jul 1995 13:49:09 -0700  
To: hfsig@tapr.org  
Message-ID: <199507282049.NAA28905@unix.ka9q.ampr.org>  
Rendered body type: text/plain

> To help the argument along, just for starters consider the  
> following: The impact on other nearby stations for a digital  
> modulated 100W packed into 500 Hz (0.2W/Hz) compared to  
> 100W spread over 3KHz (0.033W/Hz). Even with the best filters,  
> how wide will a high-powered signal really be? The main concern

It may not be 100W spread over 3KHz, it might be much less. One of the advantages of going wider is the ability to use LESS total power. So you can often reduce W/Hz twice, once by going wider and again by using less total power.

> Another example: Would the principle ideas as used in the cellular  
> phone market have worked if it was based on narrow-band  
> transmission?

No. There are good reasons, aside from simplicity, why AMPS chose FM over SSB. There are even better reasons for going wider, as with CDMA.

Phil

---

### 64. Re: [HFSIG:487] Re: FCC Regulations for HF digital

From: Phil Karn <karn@unix.ka9q.ampr.org>  
Date: Fri, 28 Jul 1995 13:56:01 -0700  
To: hfsig@tapr.org  
Message-ID: <199507282056.NAA28909@unix.ka9q.ampr.org>  
Rendered body type: text/plain

>So - we should think about talking to the  
>TAPR FCC Reg committee and some of the people at the league about this issue  
>so that they can be aware of it. I would think that Paul Rinaldo is aware of it  
>>n-- the reason that many of the alternate protocols/modes were published  
>in the last Digital Communications Conference Proceedings was to be able to  
>have a reference to then include under some 'misc' section of the rules.

>I forget all the maneuvers required. Phil is on the Future Systems  
>Committee and might be able to comment....??

Most of the members of the Future Systems committee are already sold on this stuff. So is the FCC, for that matter -- they've shown a remarkable openness to rule changes that promote technical progress.

The problem, in my opinion, lies with the average ham and with the ARRL leadership who represents them. This has been very apparent in the recent attempts to liberalize the SS rules. They don't understand the technology and its benefits, and as often happens people fear what they don't understand.

The key here is education, backed up by convincing demonstrations of the new technology in action. STAs can be easily had if they're necessary.

Phil

---

### 65. Re: [HFSIG:503] Re: FCC Regulations for HF digital

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Sat, 29 Jul 95 23:05:42 -0500  
To: hfsig@tapr.org  
Message-ID: <m0scPdH-0002AeC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

In Phil's message of 29 Jul 1995 at 1734 CDT, he writes:

> >So - we should think about talking to the  
> >TAPR FCC Reg committee and some of the people at the league about this issue  
> >so that they can be aware of it. I would think that Paul Rinaldo is aware of it

> >n-- the reason that many of the alternate protocols/modes were published  
> >in the last Digital Communications Conference Proceedings was to be able to  
> >have a reference to then include under some 'misc' section of the rules.  
>  
> >I forget all the maneuvers required. Phil is on the Future Systems  
> >Committee and might be able to comment....??  
>  
> Most of the members of the Future Systems committee are already sold  
> on this stuff. So is the FCC, for that matter -- they've shown a remarkable  
> openness to rule changes that promote technical progress.  
>  
> The problem, in my opinion, lies with the average ham and with the  
> ARRL leadership who represents them. This has been very apparent in  
> the recent attempts to liberalize the SS rules. They don't understand  
> the technology and its benefits, and as often happens people fear what  
> they don't understand.  
>  
> The key here is education, backed up by convincing demonstrations of  
> the new technology in action. STAs can be easily had if they're necessary.

\*\*\*\*\* IMHO, the "convincing demonstration" is the key to acceptance. \*\*\*\*\*

73, Walt

PS, sorry of the additional bandwidth of the above but felt it had to  
be read before reading my comments. -- k5yfw

## 66. DWAIT/DTIME/T2 KISS parameter number

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 31 Jul 95 10:13:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <301C92BA@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

Hello everyone,

in testing Pawels multitone modem we came across a problem with the DSP4 OS,  
Leonid. It does  
not support the DTIME/DWAIT/T2 parameter. Jarkko promised to code it into  
Leonid but we do not  
have any documents on this feature. The jamming sequence of the modem makes  
the rx fall out of  
synnc which results in immediate ack. if we use maxframe > 1 the ack is  
transmitted on the following  
I frame, creating collision.

The parameter that we need is the one that adds a wait-time before \*every\*  
transmission. Now  
Leonid works so, that if there is no carrier on the frequency when it wants  
to transmit, it transmits  
immediately. Only if there is a carrier on at the moment of attempted  
transmission it starts the  
persistence / slottime lottery. Today most TNCs have a parameter which  
defines how long to  
wait everytime before transmission. This is called DWAIT/DTIME or T2.  
Correct me if I am wrong.

What is the KISS parameter number for this parameter? It is needed in order  
to make Leonid  
compatible with existing KISS drivers like TFKISS and TFPCR. If anyone of  
you have an updated  
list of the KISS parameters it would be more than appreciated.

We have managed to make quite good QSO's with the 9 carrier multitone modem.  
The original 15  
carrier version did not work well, but the 9 carrier version works ok. We  
could not make any reasonable  
file transfer tests because of this KISS parameter lacking. We had a long  
ragchew using the modem  
and could verify that it except this KISS problem it worked quite ok. No  
results are available on the  
bps speed yet. When we can transmit multiple packets we will run several  
file transfer tests in order  
to calculate real transmission speed. We also plan to make more tests on  
longer distances from Jonis  
summer cottage and using the crowded lowbands, when we get a fully operating  
modem.

The 9QPSK modem is according to our tests a fairly good working model. It is sensitive to distortion, but works fairly well with low modulation levels. There is a annoying brum on the TX line when not transmitting which originates from the PSK software, not the electronics. This could be checked out by someone who knows the software. It is annoying when we use the same rigs for talkback, as the hum is coming trough to the mic side and to the SSB audio.

We will be back with more tests and results after the Leonid modification.

The test equipments we use are in the both ends DSP4 and FT757GX TRX with vertical antennas. Power has been about 30 W. We have been working on 29.100 MHz SSB and FM. The distance between me and Joni, OH2NJR is about 10 km.

73, Timo OH6KK/OH2LVZ

---

## 67. Re: [HFSIG:505] DWAIT/DTIME/T2 KISS parameter number

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 31 Jul 1995 08:45:29 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <6CF49693604@frl.orst.edu>  
Rendered body type: text/plain

Dear Timo,

Outstanding! Keep us informed on how things progress.

I am busy looking into improving the transmitted pulse shaping. This probably will clean up spectral leakage and at the same time help a bit on tuning the signal. Some form of tuning indication would help things too.

Keep up the good work - I am sure Pawel will be delighted to hear the news when he returns after vaction.

Good work with the 9-tone version, Frode. Do you think that it is the SSB filters causing the problem with the 15-tone version?

73's

--Johan, KC7WW

---

# August 1995

Original source file: hfsig.9508

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-08/TAPR-HF\\_1995-08.txt](https://archive.org/download/TAPR-HF_1995-08/TAPR-HF_1995-08.txt)

## 1. Re: [HFSIG:505] DWAIT/DTIME/T2 KISS parameter number

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 1 Aug 1995 10:03:18 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <6E895C81979@frl.orst.edu>  
Rendered body type: text/plain

Hi all,

I am not sure that everyone picked up Timo's question and request for help?

We need to know some information on the KISS protocol: does anyone know what the KISS parameter (code) for T2/Dwait/Dtime is? Who might know? Any help will be greatly appreciated.

Thanks in advantage.

--Johan, KC7WW

---

## 2. Re: [HFSIG:507] Re: DWAIT/DTIME/T2 KISS parameter number

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Tue, 1 Aug 95 17:21:25 -0500  
To: hfsig@tapr.org  
Message-ID: <m0osdPgj-0002AnC@sacdm10.kelly.af.mil>  
Rendered body type: text/plain

In your message of 1 Aug 1995 at 1212 CDT, you write:

> Hi all,  
>  
> I am not sure that everyone picked up Timo's question and request for help?  
>  
> We need to know some information on the KISS protocol: does anyone know  
> what the KISS parameter (code) for T2/Dwait/Dtime is?  
> Who might know? Any help will be greatly appreciated.  
>  
> Thanks in advantage.  
>  
> --Johan, KC7WW  
>  
>  
>

I'm not at home now so can't get the information but Timo might try sending the request to nos-bbs@hydra.carleton.ca or tcp-group@ucsd.edu

73, Walt/K5YFW

---

## 3. Re: [HFSIG:508] Re: DWAIT/DTIME/T2 KISS parameter number

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 1 Aug 1995 16:38:14 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <6EF2B78637E@frl.orst.edu>  
Rendered body type: text/plain

Hi Walt,

Good idea.

I don't think either Timo or I am subscribed there - could you do us a favor pse: cross-post and forward any feedback?

Thanks much.

--Johan

>In your message of 1 Aug 1995 at 1212 CDT, you write:

>> Hi all,  
>>  
>> I am not sure that everyone picked up Timo's question and request for help?



>>  
>> We need to know some information on the KISS protocol: does anyone know  
>> what the KISS parameter (code) for T2/Dwait/Dtime is?  
>> Who might know? Any help will be greatly appreciated.  
>>  
>> Thanks in advance.  
                  ^^^^^^^^^ thats a new one - hi.  
  
>>  
>> --Johan, KC7WW  
>>  
>>  
>>  
  
>I'm not at home now so can't get the information but Timo might try  
>sending the request to nos-bbs@hydra.carleton.ca or tcp-group@ucsd.edu  
>  
>73, Walt/K5YFW

---

#### 4. Re: [HFSIG:509] Re: DWAIT/DTIME/T2 KISS parameter number

From: k5yfw@sacdm10.kelly.af.mil (WALT DUBOSE - K5YFW)  
Date: Wed, 2 Aug 95 07:52:37 -0500  
To: tcp-group@ucsd.edu  
Cc: hfsig@tapr.org  
Message-ID: <m0sddHp-000299C@sacdm10.kelly.af.mil>  
Rendered body type: text/plain  
  
Could someone answer Timo's question and send the reply to:  
  
          hfsig@tapr.org  
  
Thanks,  Walt/K5YFW

In Timo's message of 1 Aug 1995 at 11212 CDT, he writes:  
>  
> I need to know some information on the KISS protocol: does anyone know  
> what the KISS parameter (code) for T2/Dwait/Dtime is?  
> Who might know? Any help will be greatly appreciated.  
>  
> Thanks in advance, Timo

---

#### 5. re: DWAIT and KISS

From: Mike Curtis <wd6ehr@kaiwan.com>  
Date: Wed, 2 Aug 1995 10:57:10 -0700 (PDT)  
To: hfsig@tapr.org  
Message-ID: <199508021757.KAA13871@kaiwan009.kaiwan.com>  
Rendered body type: text/plain  
  
There is no DWAIT in KISS (fortunately!) It has been replaced by persist  
and slotttime, which is vastly superior to the old DWAIT parameter. Also,  
Ppersist is incompatible with DWAIT.  
  
-- mike

---

#### 6. FFT speeds?

From: Phil Karn <karn@qualcomm.com>  
Date: Wed, 2 Aug 1995 15:07:45 -0700  
To: hfsig@tapr.org  
Message-ID: <199508022207.PAA29619@servo.qualcomm.com>  
Rendered body type: text/plain  
  
I wasn't happy with any of the FFT routines I found on the net or in  
textbooks, so over the weekend I wrote up one in C. It does a 256-point  
double precision complex FFT in 870 microseconds on the Pentium 90  
and 3.34 milliseconds on the AMD 486DX4-100. Anybody have some comparative  
speed figures?  
  
Phil

---

#### 7. Re: [HFSIG:512] FFT speeds?

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Wed, 2 Aug 1995 16:06:16 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <706A3735912@frl.orst.edu>  
Rendered body type: text/plain

Phil,

> I wasn't happy with any of the FFT routines I found on the net or in  
> textbooks, so over the weekend I wrote up one in C. It does a 256-point  
> double precision complex FFT in 870 microseconds on the Pentium 90  
> and 3.34 milliseconds on the AMD 486DX4-100. Anybody have some comparative  
> speed figures?  
>  
> Phil

Could you tell us whether this is floating point or integer, and what  
radix ? Sounds very impressive. Just for sake of comparison,  
the 12 MHz ADSP-21xx benchmark for a Radix 4, 256 point integer FFT is  
590 us. The 1024 point R4 checks in at 2.98 ms. Radix 2 is nearly double  
those figures. All these are complex, 16-bit data words.

I am sure that you will get a reasonable answer if you post this question  
to COMP.DSP - this is a popular topic.

--Johan

---

## 8. re: DWAIT and KISS

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>  
Date: Thu, 3 Aug 1995 09:01:04 +0200 (MESZ)  
To: hfsig@tapr.org  
Message-ID: <199508030701.JAA18701@hades>  
Rendered body type: text/plain

>  
> There is no DWAIT in KISS (fortunately!) It has been replaced by persist  
> and slottime, which is vastly superior to the old DWAIT parameter. Also,  
> Ppersist is incompatible with DWAIT.  
>  
>  
> -- mike  
>

Actually the problem (at least with some implementations of KISS on TNC2)  
is, that they do not start with the slottime. Depending on the  
persistence setting these KISS versions start to transmit immediately  
after DCD is gone.  
Better would be to wait at least one time slottime before starting  
the persistence algorithm.

Alexander

--  
\*-----+-----\*  
| Alexander F. Kurpiers | |  
| Institut f. Uebertragungstechnik | Voice: +49-6151-162369 |  
| u. Elektroakustik | Fax : +49-6151-165545 |  
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de |  
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0eam.#hes.deu.eu |  
\*-----+-----\*

---

## 9. T2/Dwait/Dtime

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Wed, 02 Aug 95 09:25:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>, "'TAPR-TNC'" <tapr-tnc@tapr.org>  
Message-ID: <3020A5AE@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

Hello,

I checked some documents out and found that Dtime and Dwait are one  
parameter and T2 is another. T2 simply seems to be the parameter which is  
called Resptime in for instance the G8BPQ manual.

Dwait and Dtime is according to the TheFirmware 2.6 manual a parameter that  
puts a waiting time between the moment when the transmission is intended to  
start to the moment when it is really starting. This is the parameter for  
which I would need the KISS parameter code.

According to the TF manual dwait/dtime is using the value of one timeslot of  
the persistence algorithm. Thus the modification consist of simply adding  
one timeslot before every transmisson. I have seen this time beeing  
independently adjustable in some applications, so I would like to be sure  
about how the parameter really works and to get the KISS code for it, if

there is one.

Timo

---

## 10. Robust modems

From: Danny Higgins <Danny@edstks1.demon.co.uk>  
Date: Thu, 03 Aug 1995 17:32:23 GMT  
To: hfsg@tapr.org  
Message-ID: <229@edstks1.demon.co.uk>  
Rendered body type: text/plain

Hi, Johan, and many thanks for pointing me at HFSIG.  
I've downloaded the discussions for the past year, and I have a lot of reading to do!

Let me introduce myself. I have been active on HF for 27 years now, mainly CW, and my main interest is chasing DX with modest power and a small garden. I have never owned a linear or a beam but I wouldn't call myself a QRP fanatic. My interest is in robust, reliable HF communications, both professionally and at home. This has tended towards robust slow speed data, not high speed data transfer which is available via internet, etc.

For many years now I have been interested in coherent CW as a robust HF modulation system. Over the years I have been involved in several projects at work which could lend themselves to improving the CCW waveform.

Basic CCW requires 100 milliseconds for a dot, 300 milliseconds for a dash, with the gaps between dots and dashes as exact multiples of 100 milliseconds, giving a Morse speed of 12 WPM. The difficulty with this is the frequency accuracy required and the synchronisation between both ends of the link. In practice, a 9Hz wide non-ringing integrate and dump filter can be used effectively to detect the tones once the signal has been found. The integrate and dump filter compares its output at the end of every dot period with the no-signal condition to determine if the carrier was present.

Imagine now the effect of sending a different tone for the key-up condition. This signal would be 10Hz away from the key-down signal (assuming a 100 millisecond dot length, since the tones must be orthogonal). The key-up signal is effectively another CCW signal which can be detected in the same way, but now the data decision is based on the difference between the key-up and key-down tone detectors. This must give a 3 dB improvement in signal to noise ratio at the expense of another coherent tone detector (free if you are using DSP!). This is a 2 tone version of the 32 tone Piccolo system.

There is still the problem of synchronisation. Instead of having one tone for key-up and another tone for key-down, suppose that a pair of tones are used, each lasting 50 milliseconds and they must now be 20Hz apart to retain orthogonality. Let the tones be called Mark (M) and Space (S). Let a DOT be represented by "SS" and a DASH be represented by "MM". The key-up condition can be represented by:

"MSMSMSMSMS...."

The sequence "SM" is invalid and shows that either an error has occurred or the system is out of sync.

The receiving system can automatically synchronise to a stream of "...MSMSMS..." tones at the start of a transmission. Three detectors can be used at the decoder, one sampling early, one in the middle and one sampling late. The relative outputs of these detectors can be used to automatically advance and retard the sampling signals to achieve synchronisation (again this is cheap in DSP but expensive in a custom hardware solution).

Since a DASH contains no more information than a DOT there is no reason for it to be 3 times longer. The gap between DOTS and DASHES also does not need to be sent, since the detector can uniquely identify them. The word "PARIS" for example would be:

"SSMMMMSSSSMMMMSSSSSSSSSSSSSSSSSSSS"

using "MS" to define the boundary between individual letters. This coding gives an effective 36 WPM based on the current Morse alphabet and a 100 millisecond tone pair sequence. In practice this can be improved further still because the original Morse alphabet was devised with a DASH being 3 times longer than a DOT. In this system the time taken to send the letter "I" is the same as the time taken to send to letter "M", so a more optimum coding system could be devised based on the frequency of use of all characters (including numbers). The complexity of the code doesn't matter because it will all be done by the machine. The code could also be extended to have short unique sequences for commonly used combinations of characters (e.g. QTH, QSY, 599, etc.). A speed in excess of 40 or 50 WPM could easily be achieved and the

50 millisecond tone length, which is a long time on HF, is much longer than any path delays therefore there should be no inter-symbol interference. This system is basically a 2 tone version of 6 tone Piccolo (LA1117).

Having produced a code with 20Hz shift, it is possible to chop this with a pseudo random sequence (direct sequence spread spectrum) in such a way that the signal occupies a 3 kHz bandwidth. It is theoretically possible to obtain about 30 dB of signal to interference rejection this way, on top of the processing gain already achieved by coherent detection. Different QSOs could co-exist in the same channel if different spreading codes are used, or narrowband signals could be retained for spectrum efficiency (it is also easier to hear someone calling CQ).

Error detecting and correcting codes could also be applied for additional robustness.

I have worked for 3 UK communications companies, two using Piccolo on HF and the other using narrowband (<3 kHz) direct sequence spread spectrum on HF. It would be nice to combine the advantages of both.

This note is very similar to one sent to Jim Mortenson, N2HOS, at the American Digital Radio Society (to which I now subscribe) and may be printed in the July issue.

On the subject of DSP platforms, I have just purchased a P38 DSP modem card from HAL Communications, and have the EVM5X evaluation module for the TMS320C50. Since the software is all downloadable from the PC, it should be possible for someone to implement something like this to run on existing hardware and therefore get a lot of people up and running fast with these new modes. This was always the problem with the old CCW system, since it required people to build filters which were inflexible. What about a stand-alone box with a morse input and output which would plug into a rig and not require a computer?

On the subject of external frequency standard inputs for amateur receivers - beware that some rigs use an internal standard for the synthesizer, then mix it with a free running crystal oscillator to get it on the right band!

On the subject of time standards - yes a cheap MSF (a UK 60 kHz time standard transmission) receiver can be built and software exists to take the demodulated time signal through a pin on the printer socket to automatically update the PC clock. (This can then be used in a system to monitor the international chirpsounder network on quiet spot frequencies using a matched DSP filter to identify particular emitters by their time/frequency offset and provide real-time frequency propagation information).

Keep up the good work!!

73s for now, de Danny Higgins, G3XVR

--  
Danny Higgins

---

## 11. Re: [HFSIG:509] Re: DWAIT/DTIME/T2 KISS parameter number

From: Joerg Reuter <jreuter@lykos.tng.oeche.de>  
Date: Wed, 2 Aug 1995 17:36:36 +0000 (GMT)  
To: WALT DUBOSE - K5YFW <k5yfw@sacdm10.kelly.af.mil>  
Cc: hfsig@tapr.org  
Message-ID: <Pine.LNX.3.91.950802170125.3628A@lykos.tng.oeche.de>  
Rendered body type: text/plain

On Wed, 2 Aug 1995, WALT DUBOSE - K5YFW wrote:

> Could someone answer Timo's question and send the reply to:  
>  
> hfsig@tapr.org

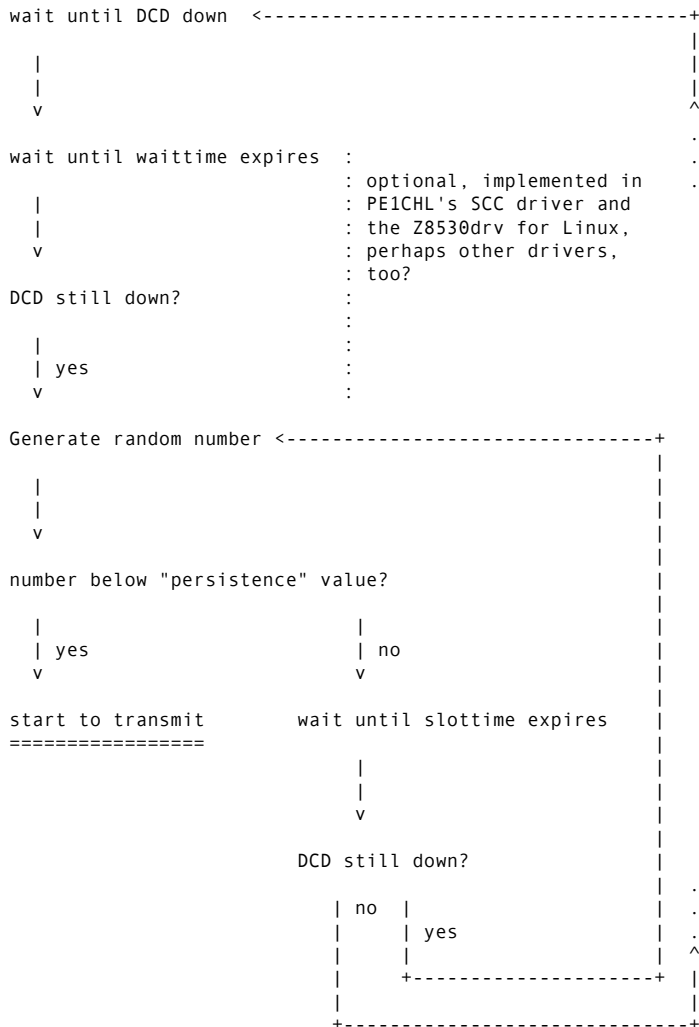
Is this another mailing list?

>  
> Thanks, Walt/K5YFW  
>  
> In Timo's message of 1 Aug 1995 at 11212 CDT, he writes:  
> >  
> > I need to know some information on the KISS protocol: does anyone know  
> > what the KISS parameter (code) for T2/Dwait/Dtime is?  
> > Who might know? Any help will be greatly appreciated.  
> >

> > Thanks in advance, Timo

T2 is an AX.25 Level 2 parameter, not a KISS parameter. If this timer expires, the whole received packet will be acknowledged through a RR. A too small T2 will result in an frame-per-frame ack. (with TheFirmware below version 2.6 you will sometimes see cascades of RR-frames after a frame if you set T2 too low: RR1 / RR2 / RR3 / ...)

DWait --- You probably think of the time the TNC waits after DCD went down before it starts to transmit. This is done with the P-Persistence (sp?) / Slottime algorithm.



KISS usually supports at least the following commands:

|                       |                         |
|-----------------------|-------------------------|
| 0 - Data              | (of course!)            |
| 1 - Tx delay          | (timer)                 |
| 2 - persistence       | (value)                 |
| 3 - slottime          | (timer)                 |
| 4 - tx tail           | (timer)                 |
| 5 - fullduplex        | (flag)                  |
| 255 - reset KISS mode | (or return to Hostmode) |

all timers in 1/10 s

Joerg Reuter      ampr-net: dl1bke@db0pra.ampr.org  
AX-25      : DL1BKE @ DB0LJ.#RPL.DEU.EU  
Internet: jreuter@lykos.tng.oche.de

## 12. Remez exchange algorithm code?

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 3 Aug 1995 13:42:05 -0700

To: hfsig@tapr.org  
Message-ID: <199508032042.NAA08011@servo.qualcomm.com>  
Rendered body type: text/plain

Does anybody have a machine-readable copy of the Remez exchange algorithm code for designing FIR filters? There's a FORTRAN listing in Rabiner & Gold that I'd like to avoid typing in (and translating to C) if possible. SOMEBODY out there must have it, right?

Phil

---

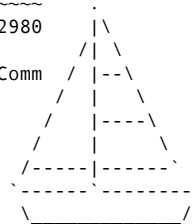
### 13. Re: [HFSIG:516] Robust modems

From: paul@paccomm.com  
Date: Thu, 03 Aug 95 17:50:10 UTC  
To: hfsig@tapr.org  
Message-ID: <2698@paccomm.com>  
Rendered body type: text/plain

CCW timing is easy....  
derive your timing from GPS time. You'll then be accurate to around 40ns for the best receivers or a more typical value of 250ns.....

~~~~~  
Paul Evans, W4/G4BKI paul@paccomm.com +1 (813) 874-2980  
Fax:+1 (813) 872-8696  
Views expressed here are not necessarily those of PacComm  
Don't surf the net, SAIL it!  
Captain of S/V "Spindrift", Dunedin, FL.  
Tall rig (58'), Winged Keel (4' 2"), full batten main

PacComm Packet Radio Systems  
4413 N. Hesperides St., Tampa, FL 33614-7618



---

### 14. Re: [HFSIG:518] Remez exchange algorithm code?

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Thu, 03 Aug 1995 18:22:09 -0800  
To: hfsig@tapr.org  
Message-ID: <9508040208.AA08515@ucs.orst.edu>  
Rendered body type: text/plain

>Does anybody have a machine-readable copy of the Remez exchange  
>algorithm code for designing FIR filters? There's a FORTRAN listing in  
>Rabiner & Gold that I'd like to avoid typing in (and translating to C)  
>if possible. SOMEBODY out there must have it, right?  
>  
>Phil  
>

I uploaded a package to ftp.tapr.org tapr/SIG/hfsig/upload/remez.zip that you might want to look at. It contains both 16 and 32-bit versions including C source code. The author is Egil Kvaleberg and I think he did a nice job.

I have a Fortran version as well - it may well be the one in Rabiner and Gold, if you are interested in that, let me know.

--Johan

---

### 15. KISS Info received, Thank you.

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Fri, 04 Aug 95 10:05:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>, "'TAPR-TNC'" <tapr-tnc@tapr.org>  
Message-ID: <3021D6E2@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

Hello everybody!

I have received awhole bunch of information concerning KISS. Thank you very much  
for putting time on this! We have now made the modification to Leonid and Beta-testing  
is going on.

Thus there is no need for more replys.

Thank you very much, all the help from everyone was very much appreciated.

Timo OH6KK/OH2LVZ

## 16. RE: [HFSIG:517] Re: DWAIT/DTIME/T2 KISS parameter number

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Fri, 04 Aug 95 10:06:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <3021D6EB@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

>T2 is an AX.25 Level 2 parameter, not a KISS parameter.

I didn't know that before, but I do know it now! Thanks to everyone who bothered answering!

>I A too small T2 will result in an frame-per-frame  
>ack. (with TheFirmware below version 2.6 you will sometimes  
>see cascades of RR-frames after a frame if you set T2 too low:  
>RR1 / RR2 / RR3 / ...)

To me it seems like most KISS TNC's do this anyway.

Timo

---

## 17. Re: [HFSIG:513] Re: FFT speeds?

From: Phil Karn <karn@unix.ka9q.ampr.org>  
Date: Fri, 4 Aug 1995 03:34:37 -0700  
To: FORRERJ@frl.orst.edu  
Cc: hfsig@tapr.org  
Message-ID: <199508041034.DAA08159@unix.ka9q.ampr.org>  
Rendered body type: text/plain

>Could you tell us whether this is floating point or integer, and what  
>radix ? Sounds very impressive. Just for sake of comparison,

Floating point -- double precision. The butterflies are all radix 2, though I do optimize out the complex multiplies in the cases where the twiddle factors are 1 and -j. Avoiding the complex multiply in the latter case is what gives the radix 4 FFT most of its advantage.

Wednesday night I used it in a bandpass filter that works by fast convolution with overlap+add. It's a hack I needed to filter some test audio that I'm working into a digital voice demo to complement the one I already did for CW vs PSK with FEC. The idea is to produce a synthetic noisy SSB signal that uses the same satellite transponder energy as a vocoded speech signal at some data rate (either 13kb/s for the GSM coder or 8kb/s for the Qualcomm QCELP coder) over a modem with some specified EB/N0.

The big question in any power efficiency comparison of digital voice to SSB is the target audio S/N ratio to be used in the comparison. If a high S/N ratio is required, the digital scheme is clearly the winner. But if people are willing to tolerate very low SSB S/N ratios, it's harder for the digital scheme to save power. And I have a rather poor quantitative feel for what various SSB S/Ns really sound like, hence my experiments.

Phil

---

## 18. Re: [HFSIG:516] Robust modems

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Fri, 4 Aug 1995 08:30:53 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <72F0D5C7F2D@frl.orst.edu>  
Rendered body type: text/plain

Hi Danny,

First - let me welcome both you and Paul Evans to HFSIG. We are looking forward sharing in your ideas and talents.

I have had several requests lately from folks that read this SIG asking for more basic materials at a level for the average ham. This is something that I intend doing for future HFSIG columns in PSR, however, once I have a bit more free time - so please be patient in the mean time. That brings me to CCWW:

CCW, in my opinion, is probably one of the neatest ways of learning DSP from the ground up. Especially now with these low-cost EVM's, and all those DSP-93's out there, maybe the time is ripe for someone like you to develop some DSP code and write about it so that we can get more folks involved and participating.

I am aware of at least two CCW programs out there - one is based on the work of Bill de Carle, VE2IQ (?) that runs on a PC and uses his very simple sigma-delta A/D convertor (described in QST and QEX). The output of this program is computer-generated tones - the operator still has to decode the CW by ear though. This would be a great first step to learn DSP principles - mostly, common sense is involved (assuming the operator can decode CW by ear). Then there is another CCW program by a DL ham (sorry I don't recall his name) that takes this another step further - it actually decodes and prints the CW.

Your suggestion of a two-tone signalling system sounds intriguing - however, we loose the ability to decipher the CW by ear. Hold the thought for the moment - I like your idea for extending CCW to a narrow-band SS system. This may perhaps be the breakthrough that we are waiting for! How does it sound, if after you get regular CCW working, implementing the SS feature. Regular narrow-band CCW is used to call CQ, synchronise and establish the link - the mode is then switched over to SS. The main requirement is that it must be possible for monitoring the transmission by a third party - I am sure that with some ingenuity, this may be possible. It probably will require that US amateurs apply for STA to participate, but wouldn't that be a wonderful opportunity to get exposed to these contemporary concepts?

Good luck with your project. Keep us posted on your progress and let us know how we can help.

73's

--Johan, KC7WW

---

## 19. Re: [HFSIG:519] Re: Robust modems

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 4 Aug 1995 15:20:02 -0700  
To: paul@paccomm.com  
Cc: hfsig@tapr.org  
Message-ID: <199508042220.PAA17024@servo.qualcomm.com>  
Rendered body type: text/plain

>CCW timing is easy....  
>derive your timing from GPS time. You'll then be accurate to around  
>40ns for the best receivers or a more typical value of 250ns.....

Except, of course, for the effects of signal propagation delay...

Phil

---

## 20. MIL-STD-188-141A

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Sun, 06 Aug 1995 05:39:48 +0100  
To: hfsig@tapr.org  
Message-ID: <199508060545.AA22825@felix.dircon.co.uk>  
Rendered body type: text/plain

If anyone is interested there is a postscript copy of the ALE mil standard and a final draft of the proposed H.F data link protocol for use with FS1052 and some psk code and even some viterbi stuff on....

ftp://atanasoff.nmsu.edu/pub/hf

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 21. FEC packages updated

From: Phil Karn <karn@unix.ka9q.ampr.org>  
Date: Sat, 5 Aug 1995 23:06:48 -0700  
To: hfsig@tapr.org, tcp-group@ucsd.edu  
Message-ID: <199508060606.XAA10906@unix.ka9q.ampr.org>  
Rendered body type: text/plain

I've put out updated versions of the Fano and Viterbi decoder packages I originally released in March.



The changes aren't extensive; mostly minor bug fixes and some cosmetic reworking. They're described in the new readme files. My web page

<http://www.qualcomm.com/people/pkarn/ham.html>

has been updated to point to the updated files.

Note: since the files themselves reside in ftp.ucsd.edu's anonymous upload directory, I was unable to overwrite or delete the old versions. So I had to create new file names. Flush any old copies you may have of the web page mentioned above.

Phil

---

## 22. Re: [HFSIG:524] Re: Robust modems

From: Danny Higgins <Danny@edstks1.demon.co.uk>  
Date: Fri, 11 Aug 1995 12:38:00 GMT  
To: hfsig@tapr.org  
Message-ID: <236@edstks1.demon.co.uk>  
Rendered body type: text/plain

In message <72F0D5C7F2D@frl.orst.edu> hfsig@tapr.org writes:  
Hi again Johan

> CCW, in my opinion, is probably one of the neatest ways of learning DSP  
> from the ground up. Especially now with these low-cost EVM's, and all those  
> DSP-93's out there, maybe the time is ripe for someone like you to develop  
> some DSP code and write about it so that we can get more folks involved and  
> participating.

This is the only reason that I want to get into DSP. I am way down the learning curve, but I've got the kit and the enthusiasm, all I need now is time and knowledge!!

> I am aware of at least two CCW programs out there - ....

I am aware of these programs, but as you can see I want to take it further.

> Your suggestion of a two-tone signalling system sounds intriguing -  
> however, we loose the ability to decipher the CW by ear.

What I suggested was a modem with a morse key input and loudspeaker output which sent the reconstituted morse back to the operator, using a FIFO to balance out the speed differences. With about 50 WPM capability, I reckon the modem would spend most of its time idling!

> Hold the thought for the moment -  
> I like your idea for extending CCW to a narrow-band SS  
> system. This may perhaps be the breakthrough that we are waiting for! How  
> does it sound, if after you get regular CCW working, implementing the SS  
> feature. Regular narrow-band CCW is used to call CQ, synchronise and  
> establish the link - the mode is then switched over to SS. The main  
> requirement is that it must be possible for monitoring the transmission by  
> a third party - I am sure that with some ingenuity, this may be possible.  
> It probably will require that US amateurs apply for STA to participate, but  
> wouldn't that be a wonderful opportunity to get exposed to these  
> contemporary concepts?

The Piccolo modem by Adrian Nash would form the basis for this type of CCW system if it only used 2 tones. I'm trying to get in touch with him since he is transporting it to the TMS320C50 which I use. Alternatively, why not keep all 6 tones and still apply direct sequence spread spectrum?

The spreading code would be chosen for its autocorrelation properties, not as a means of "encrypting" data. The codes would therefore be in the public domain. Experimental licences have been granted over here, but mainly for VHF work. I see no reason for not applying this to HF, as long as the signal was band limited to about 2 or 3 kHz, therefore not requiring a special IF filter in the receiver/transmitter. I wonder how long it will be before Ham rigs will generally have DSP IF filters to get round some of these problems.

--

73s de Danny Higgins, G3XVR

---

## 23. Re: [HFSIG:525] Re: Robust modems

From: Danny Higgins <Danny@edstks1.demon.co.uk>  
Date: Fri, 11 Aug 1995 12:52:41 GMT  
To: hfsig@tapr.org  
Message-ID: <237@edstks1.demon.co.uk>  
Rendered body type: text/plain

Hi, Phil.

>CCW timing is easy....  
>derive your timing from GPS time. You'll then be accurate to around  
>40ns for the best receivers or a more typical value of 250ns.....  
>  
> Except, of course, for the effects of signal propagation delay...  
>  
> Phil  
>

You could end up paying more for GPS than for the modem! I would hope to build a small stand alone modem for use with any transceiver.

As someone else pointed out before, if you use GPS timing then you could use the propagation delay to select (or reject!) signals from particular areas.

--  
73s de Danny Higgins, G3XVR

---

## 24. Re: [HFSIG:519] Re: Robust modems

From: Danny Higgins <Danny@edstks1.demon.co.uk>  
Date: Fri, 11 Aug 1995 12:34:42 GMT  
To: hfsig@tapr.org  
Message-ID: <235@edstks1.demon.co.uk>  
Rendered body type: text/plain

In message <2698@paccomm.com> hfsig@tapr.org writes:  
> CCW timing is easy....  
> derive your timing from GPS time. You'll then be accurate to around  
> 40ns for the best receivers or a more typical value of 250ns.....

Fine, but you will spend more on GPS than I intended to on the modem!  
I would like to make this an external box which went with any transceiver.

--  
Danny Higgins

---

## 25. RE: [HFSIG:528] Re: Robust modems

From: "Sadowski,P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>  
Date: Fri, 11 Aug 95 07:18:00 cdt  
To: hfsig <hfsig@tapr.org>  
Message-ID: <302B4A05@mailgtwy.nellis.af.mil>  
Rendered body type: text/plain

The comments about the CCW and SS were very interesting (exactly the kind of thing I was hoping to see in this SIG).. I'm looking for info on a

very similar topic, CCW but extremely narrow bandwidth - - 10 Hz max.. I realize the symbol rate will be slower - but my interest is in sending a few "canned" msgs - so an appropriate coding scheme can "compress" a heck of a bunch with only a few fixed messages. Mind you, I'm not interested in giving up free text altogether... just that I can accept serious compromise for free text (canned msgs are 95% of need). I'm going to show my ignorance here... but what might be the tradeoffs I can expect by using SS with low symbol rate vs CCW at low rate? Technology wise, error rate, propagation anything else???

Thanks in advance  
AL  
AH6LS  
sadowskp@usafcrqs.nellis.af.mil

---

## 26. Re: [HFSIG:531] Re: Robust modems

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 11 Aug 1995 10:03:59 -0800  
To: hfsig@tapr.org  
Message-ID: <9508111749.AA00474@ucs.orst.edu>  
Rendered body type: text/plain

Dear Al,

>

>The comments about the CCW and SS were very interesting (exactly the  
>kind of thing I was hoping to see in this SIG)..

I agree - if this kind of approach is taken, it would be a breakthrough I  
believe.

>I'm looking for info on a

>very similar topic, CCW but extremely narrow bandwidth - - 10 Hz max..

The rationale behind CCW's narrow filter bandwidth, is that it is optimal  
(not necessarily minimal) for the symbol length. This is what the theory of  
"matched filters" is about. The price that one pays for having such a  
special filter is in the complexity in technology/algorithms that precisely  
extracts where each symbol start/end, and in addition maintain that with  
vengeance. This is where men and the mice are separated.

> I

>realize the symbol rate will be slower - but my interested is in sending a

>few "canned" msgs - so an appropriate coding scheme can "compress"

>a heck of a bunch with only a few fixed messages. Mind you, I'm not

>interested in giving up free text altogether... just that I can accept

>serious

>compromise for free text (canned msgs are 95% of need).

Not a bad idea - the "Q" code was intended to do that anyway.

>I'm going to

>show my ignorance here... but what might be the tradeoffs I can expect

>by using SS with low symbol rate vs CCW at low rate? Technology wise,

>error rate, propagation anything else???

>

>Thanks in advance

>Al

>AH6LS

>sadowskp@usafcrqs.nellis.af.mil

>

I am as ignorant, but let me have a shot at it - I'll let those more  
knowledgeable correct me please.

These are two different concepts:

CCW is a form of coherent demodulation/detection. All that is achieved is  
optimal receiving conditions for detecting a signal, given the  
signal-to-noise ratio. There is no built-in error detection/correction when  
QSB or QRM sets in (except of course that when you decipher CW by ear, one  
tend to "fill in the blanks" and thus intentionally are doing some form of ECC).  
Summarized: its strenghts is optimal signal detection for a given S/N. If  
there is spare S/N, i.e., good conditions (this probably includes conditions  
where regular CW at the same speed would be pretty tough), very low power  
can be used. There is nothing special about the emitted CW bandwidth - it is  
a regular on-off keyed waveform at the given symbol rate. If there is QRM on  
identically the same frequency, throughput goes down to zero.

SS is a signal spreading mode: Instead of having all the signal energy  
concentrated at one frequency, it gets spread out over a much wider band of  
frequencies - a couple of kilohertz or megahertz depending. The result is that  
a certain degree of redundancy enters the picture - if some portion of the  
signal gets blanked out by QRM, or perhaps by selective fading, there  
hopefully remains sufficient signal integrity to detect and demodulate the  
signal. Once the signal is demodulated into baseband (audio), one can still  
use matched filters as for CCW and enjoy the same gains.  
Summarized: Strengths lays in robustness and ability to withstand a  
substantial degree of interference. When applied wisely, many users may  
share the same spectral space, without one interfering with another. Sounds  
a bit like magic, doesn't it?

Hope that helps?

73's

--Johan, KC7WW

## 27. Re: [HFSIG:532] Re: Robust modems

From: Danny Higgins <Danny@edstks1.demon.co.uk>  
Date: Mon, 14 Aug 1995 17:32:07 GMT  
To: hfsig@tapr.org  
Message-ID: <248@edstks1.demon.co.uk>  
Rendered body type: text/plain

>  
> SS is a signal spreading mode: Instead of having all the signal energy  
> concentrated at one frequency, it gets spread out over a much wider band of  
> frequencies - a couple of kilohertz or megahertz depending. The result is that  
> a certain degree of redundancy enters the picture - if some portion of the  
> signal gets blanked out by QRM, or perhaps by selective fading, there  
> hopefully remains sufficient signal integrity to detect and demodulate the  
> signal. Once the signal is demodulated into baseband (audio), one can still  
> use matched filters as for CCW and enjoy the same gains.  
> Summarized: Strengths lays in robustness and ability to withstand a  
> substantial degree of interference. When applied wisely, many users may  
> share the same spectral space, without one interfering with another. Sounds  
> a bit like magic, doesn't it?  
>

If I can throw a little bit of light on SS -

If you start at the modulator with a clean signal, then spread it over, say a  
3 kHz bandwidth by introducing 180 degree phase changes from a pseudo random  
code sequence at a rate equal to half (?) the bandwidth required, what you  
get is a signal which has a  $\sin(x)/x$  distribution and sounds quite noiselike.

When this signal appears at the receiver it will have all kinds of QRM  
superimposed on it, and usually at a much greater amplitude. When the signal  
is de-spread by introducing 180 degree phase changes which are synchronised  
to the code sequence which generated the original SS signal, the effect is to  
concentrate all of the spread signal energy back into a single (or multiple)  
tone modulated waveform, but all of the QRM gets spread by this process, which  
gets you your S/N advantage.

Hope this simple non-mathematical description helps to explain where the S/N  
advantage comes from.

--  
73s de Danny Higgins, G3XVR

---

## 28. HFSIG at the Digital Conference

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Wed, 16 Aug 1995 13:02:55 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <8539EF744A2@frl.orst.edu>  
Rendered body type: text/plain

Greetings,

Thought I would let everyone know that we are planning a special  
meeting on the topic of HF digital communications at the upcoming Digital  
Conference.

I will prepare some materials for an update on various projects  
that the folks have been working on, in particular, the parallel OFDM HF  
modem, MFSK (Piccolo) HF modem, and various bits pieces of DSP hardware.  
Time permitting, other interesting technical materials may also be  
presented. However, there will be an opportunity and open invitation to  
bring up your favorite or other pressing matters for discussion.

Greg also tells me that there would be space for me to set up a demo showing  
off a couple of the DSP projects that have been discussed on this SIG. I  
will bring a laptop and some DSP hardware and give you the opportunity to  
see how far things have come along.

Please make a special effort to be there - looking forward seeing you.

73's

--Johan, KC7WW

---

## 29. HFSIG at the Digital Conference

From: Johan Forrer <forrerj@ucs.orst.edu>  
Date: Wed, 16 Aug 1995 14:56:31 -0700 (PDT)  
To: HFSIG@tapr.org  
Message-ID: <Pine.OSF.3.91.950816145337.17596A-100000@ucs.orst.edu>  
Rendered body type: text/plain

Greetings,

Thought I would let everyone know that we are planning a special meeting on HF digital communications at the upcoming Digital Conference.

I will prepare some materials for an update on various projects that the folks have been working on, in particular, the parallel OFDM HF modem, MFSK (Piccolo) HF modem, and various bits pieces of DSP hardware. Time permitting, other interesting technical materials may also be presented. However, this is an opportunity and open invitation to share your experiences or bring up other pressing matters for discussion.

Greg also tells me that there would be space to set up a demo showing off a couple of the DSP projects that have been discussed on this SIG. I will bring a laptop and some DSP hardware and give you the opportunity to see how far things have come along.

Please make a special effort to be there - looking forward seeing you.

73's

--Johan, KC7WW

---

### 30. Re: [HFSIG:535] HFSIG at the Digital Conference

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Fri, 18 Aug 1995 06:57:57 +0200  
To: hfsig@tapr.org  
Message-ID: <199508180457.GAA04197@foxbat.pix.za>  
Rendered body type: text/plain

>Greetings,  
>  
>Thought I would let everyone know that we are planning a special  
>meeting on HF digital communications at the upcoming Digital  
>Conference.  
>

Ai... wens ek kon julle join ! As julle enige van die conference se goed op magnetiese media gaan he, sal jy asb aan my dink ?

73 danie

---

### 31. Analog Devices ADSP 2181 Development kit...

From: muphaus@cris.com (Marv Uphaus)  
Date: Mon, 21 Aug 1995 02:43:53 -0500  
To: hfsig@tapr.org  
Message-ID: <5kD0wM82cCKc083yn@cris.com>  
Rendered body type: text/plain

Johan and All...

Has anyone seen the Analog Devices ADSP 2181 Development kit...???

There's an ad on page 10 of the August 95 issue of Circuit Cellar for the kit for \$89.00... Looks pretty nice... I have wanted a stand alone board with upload capabilities and if the code that Johan and others have already developed could be cut over to the 2181 chip, this might be just the thing to do some stand alone development... It certainly could be just the vehicle to get folks like me who aren't heavy duty programmers to take a shot at using some of this code on the air... Personally I keep the ham shack and the computer development areas separate and I like the stand alone idea...

Anyone have any feelings for this... For \$89.00 it's worth buying one just to play with...

Here's the whole poop...

ADSP 2181 chip - 33-MIPS with 32KB words of on-chip memory... An AD1847 stereo audio SoundPort codec... Has an RS-232 port and a dual op-amp for signal conditioning and an EPROM socket for on-board code... You get an assembler, linker, simulator and PROM formatter in the software category...

Can be made into a stand alone device... They even give you an RS-232 cable to go to your PC and a 110V cube to power it... It has Windows monitor program to control the whole thing...

WOW, I think I'm sending in my \$89.00 today...!!!

-- Marv...

-- For private communications, use my PGP public key --  
-- available at MIT public key server or "finger" me --  
-- muphaus@cris.com --

"I still think a MegaByte is the most Nine-Lives you can get in your mouth at one time." --Morris the Cat, sitting in front of his computer terminal.

---

### 32. Re: [HFSIG:537] Analog Devices ADSP 2181 Development kit...

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 21 Aug 1995 08:42:53 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <8C74C8A7BD6@frl.orst.edu>  
Rendered body type: text/plain

Hi Marv,

Thanks for letting us know about the EZ-lite. It certainly represents a very potent little DSP - plenty of MIPS and lots of on-chip memory. The 24-bit architecture also is somewhat easier to learn than some others.

However, perhaps you want to hold out getting one until you have seen the August QEX for some comparisons. I have been playing with the Motorola 56002-based EVM (\$150) and the article is about this unit. This past weekend I have been testing, and am most impressed, with how the high speed HF modem runs on the Motorola EVM. Pawel (SP9VRC) now has a 2500 bps (without FEC) version that looks very promising. If you are planning on attending the DCC, I am planning on showing the setup and give you a first-hand opportunity to see what it is about.

73's

--Johan

---

### 33. Re: [HFSIG:538] Re: Analog Devices ADSP 2181 Development kit...

From: muphaus@cris.com (Marv Uphaus)  
Date: Mon, 21 Aug 1995 11:09:36 -0500  
To: hfsig@tapr.org  
Message-ID: <A/KOwM82ciUR083yn@cris.com>  
Rendered body type: text/plain

Johan and All...

On Mon, 21 Aug 1995 10:58:31 -0500, you wrote:

>However, perhaps you want to hold out getting one until you have seen the  
>August QEX for some comparisons. I have been playing with the  
>Motorola 56002-based EVM (\$150) and the article is about this unit.

Too late...! I got so excited that this morning I ordered one just to play with... It's being shipped today... I hope you guys have not totally abandoned the ADSP 2115 project... At least there is some code out there to play with for a small potatoes programmer...

-- Marv...

-- For private communications, use my PGP public key --  
-- available at MIT public key server or "finger" me --  
-- muphaus@cris.com --

"I still think a MegaByte is the most Nine-Lives you can get in your mouth at one time." --Morris the Cat, sitting in front of his computer terminal.

---

### 34. Re: [HFSIG:539] Re: Analog Devices ADSP 2181 Development ki

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 21 Aug 1995 10:04:17 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <8C8A81B4001@frl.orst.edu>  
Rendered body type: text/plain

Hi Marv,

No, the ADI-2115 project is alive and well. As a matter of fact, if you show up at the DDC, I will be happy to show you some very neat software by Adrian, G4ZHZ, for the PSA soundcard. I won't let the cat out of the bag. Adrian also has passed on code to run MFSK - I will be

starting work on getting it ported to the sound card DSP in the near future. So, not to worry, there will be plenty of opportunities for you to explore.

One thing that we have to accept for the moment, is that there will be several DSP platforms - it will be a while until we will be able to present a written specification for the HF modems that is presently being developed. When that time comes, it would be necessary to cross-port between these platforms, if that is possible. That possibly may be a challenge. Judging from what I have seen going on in the high speed HF modem, it would be a tough job to match the 56K's performance - the modem is very hungry on cycles, and the benefits of the 24-bit architecture has its advantages in dynamic range. Anyway we will have to wait and see how it develops.

I will present further details of these projects at the DDC and at that time invite your inputs on some issues that are presently unfolding.

--Johan, KC7WW

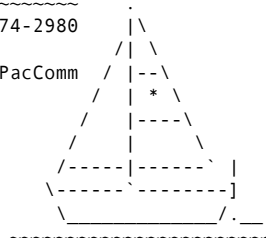
---

### 35. Re: [HFSIG:540] Re: Analog Devices ADSP 2181 Development ki

From: paul@paccomm.com  
Date: Tue, 22 Aug 95 10:38:31 UTC  
To: hfsig@tapr.org  
Message-ID: <4063@paccomm.com>  
Rendered body type: text/plain

Seems like the PTC-II team chose the right DSP to do the HF modem job..... the Mot.

~~~~~  
Paul Evans, W4/G4BKI paul@paccomm.com +1 (813) 874-2980  
Fax:+1 (813) 872-8696  
Views expressed here are not necessarily those of PacComm  
Don't surf the net, SAIL it!  
Captain of S/V "Spindrift", Dunedin, FL.  
Catalina 36: Tall rig (58'), Winged Keel (4' 2"),  
Full batten main, Boston Whaler dinghy.  
PacComm Packet Radio Systems  
4413 N. Hesperides St., Tampa, FL 33614-7618



---

### 36. Re: [HFSIG:533] Re: Robust modems

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 22 Aug 1995 14:44:22 -0700  
To: hfsig@tapr.org  
Message-ID: <199508222144.OAA15733@servo.qualcomm.com>  
Rendered body type: text/plain

An additional remark about spread spectrum processing gain -- it's a wash on a nonfading AWGN (additive white gaussian noise) channel. A spread system has no signal-to-noise advantage over a non spread system in this case.

However, SS wins big in several important real-world situations. These include fading channels, and channels with interference (intentional or benign). And it's especially helpful where there are lots of individual users with traffic that's too bursty to permit the efficient reallocation of channels or timeslots on demand.

Phil

---

### 37. Re: [HFSIG:531] Re: Robust modems

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 22 Aug 1995 15:05:18 -0700  
To: "Sadowski,P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>  
Cc: hfsig@tapr.org  
Message-ID: <199508222205.PAA15792@servo.qualcomm.com>  
Rendered body type: text/plain

>compromise for free text (canned msgs are 95% of need). I'm going to  
>show my ignorance here... but what might be the tradeoffs I can expect  
>by using SS with low symbol rate vs CCW at low rate? Technology wise,  
>error rate, propagation anything else???

In general, SS is better at extremely low rates. Extremely narrowband techniques often have problems with carrier frequency uncertainty and guard-banding, and SS can be easier to handle in practice.

Phil

---

### 38. Re: [HFSIG:543] Re: Robust modems

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 22 Aug 1995 15:38:26 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <196BA07F41@frl.orst.edu>  
Rendered body type: text/plain

Hi Phil,

Just a question out of curiosity. I gather that the STA for SS does not allow for the usage of simultaneous P-N sequences and frequency hopping schemes. Without this means of SS, would it be technically possible then to solve the near-far problem for amateur communications?

--Johan

---

### 39. Re: [HFSIG:544] Re: Robust modems

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 22 Aug 1995 21:07:31 -0700  
To: hfsig@tapr.org  
Message-ID: <199508230407.VAA16765@servo.qualcomm.com>  
Rendered body type: text/plain

Near-far is a potential problem for all spread spectrum systems, regardless of spreading method. Frequency hopping does have the advantage of making it easier to deal with hopping collisions through coding, but there are techniques applicable to direct sequence as well. IS-95 (which is direct sequence) is one example. Power control is the key.

Phil

---

### 40. Re: [HFSIG:532] Re: Robust modems

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Sat, 12 Aug 1995 05:46:30 +0100  
To: hfsig@tapr.org  
Message-ID: <199508120553.AB16802@felix.dircon.co.uk>  
Rendered body type: text/plain

>  
>CCW is a form of coherent demodulation/detection.

>  
>--Johan, KC7WW

>  
>  
>  
Hello,

From my memory the CCW filters I have seen described in the past have used Quadrature matched filters which according to my understanding of the book :

Communication Systems 3/e by Simon Haykin ISBN 0-471-30584-7 published by Wiley, are designed for non-coherent reception.

I guess this means that the phase in CCW is not necessary to know just the QRG and the start and end of the integration period ?

Back to this 8ary FSK modem I am still investigating, well I spoke to a guy at Leeds University here in the U.K who works in the H.F radio research dept and he advised me that I should be using matched filters for my 8ary modem.

So this CCW discussion ties in nicely with what I am now looking at. At least rectangular pulses are the simplest matched filter to build.

Regards

Charles

(QTH somewhere on the learning curve!)

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----



#### 41. Re: [HFSIG:546] Re: Robust modems

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Wed, 23 Aug 1995 20:36:09 +0100  
To: hfsig@tapr.org  
Message-ID: <199508232037.AA17468@felix.dircon.co.uk>  
Rendered body type: text/plain

>  
>>  
>>CCW is a form of coherent demodulation/detection.  
>>  
>>--Johan, KC7WW  
>>  
>>  
>Hello,  
> From my memory the CCW filters I have seen described in the past have used  
>Quadrature matched filters which according to my understanding of the book :

Wow,  
I sent that message about 1 week ago, like a lot of things after re-reading the books I regretted sending the message! Still I can't see why it took so long to be reflected back from the list server.

I am still playing with the matched filters. Still things have advanced at this end as I am now using PC-DSP Professional and PC-SIM which allows me to mock up the modem before committing it to DSP code.

It has a few bugs in it but appears good value for money at \$200 rather than the \$2000 + for matlab!

Hope you all enjoy the DCC I am looking forward to reading the conference notes. I have just finished reading the proceedings of HF95 that took place in Sweden last week.

Regards Charles

---

#### 42. RE: [HFSIG:547] Re: Robust modems

From: "Sadowski,P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>  
Date: Thu, 24 Aug 95 08:48:00 cdt  
To: hfsig <hfsig@tapr.org>  
Message-ID: <303C81F2@mailgtwy.nellis.af.mil>  
Rendered body type: text/plain

Please, more info on these two packages

Thanks in advance

Al AH6LS  
sadowskp@usafcrqs.nellis.af.mil  
-----

>From: hfsig  
>To: hfsig  
>Subject: [HFSIG:547] Re: Robust modems  
>Date: Wednesday, August 23, 1995 3:50PM

> PC-DSP Professional and PC-SIM

> appears good value for money at \$200 rather than  
> the \$2000 + for matlab!

> Regards Charles

---

#### 43. Re: [HFSIG:548] Re: Robust modems

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Thu, 24 Aug 1995 17:39:07 +0100  
To: hfsig@tapr.org  
Message-ID: <199508241741.AA04264@felix.dircon.co.uk>  
Rendered body type: text/plain

>  
>Please, more info on these two packages  
>  
>Thanks in advance  
>  
>Al AH6LS  
>sadowskp@usafcrqs.nellis.af.mil

```

> -----
>>From: hfsig
>>To: hfsig
>>Subject: [HFSIG:547] Re: Robust modems
>>Date: Wednesday, August 23, 1995 3:50PM
>
>> PC-DSP Professional and PC-SIM
>
>> appears good value for money at $200 rather than
>> the $2000 + for matlab!
>
>> Regards Charles
>
>
Well here are the facts, I apologise for the advert at the end but I have
had a number
of requests for information about the package.

Hello,
I hope this is of interest to you! It has some bugs in it but you can
design filters with it, play with FFTs and
most other DSP functions. With the PC-SIM I have emulated an 8ary FSK modem.
I am also going to
emulate an H.F channel simulator with it, the output can be displayed or
imported into Excell. I ordered it via
the internet and it arrived in about 1 week, it fitted in a paddy bag. You
get the disks and a large manual.

```

#### PC-DSP Professional Edition -----

The professional edition of PC-DSP is available from DSP Solutions, and provides a significant number of enhancements over the student edition. Site licenses are available for educational institutions. In addition, students with a valid student ID may upgrade to the professional edition at a discounted price. For ordering information, please see the file 'ORDER.FRM'. For site license information, see the file 'SITELICE.TXT'.

Following is a partial list of enhancements in the professional edition of PC-DSP:

- \* Protected mode support. Works with 286 or better. For most operations, discrete sequences can be as long as permitted by available memory and disk space. In the student edition, discrete sequences are limited to 1024 samples.
- \* Graphs are customizable in terms of titles, labels, and colors. Support is provided for converting graphs to several popular formats including postscript, HPGL, and Latex picture formats. This allows professional-quality printer plots to be obtained.
- \* Increased limits in filter design, analysis, and simulation. In the student edition, FIR filters are limited to 55 taps, and IIR filters are limited to order 10. The professional edition raises these limits to 256 and 50 respectively.
- \* Automatic code generation for IIR and FIR filters. Assembly language code for designed filters can be automatically generated for Motorola DSP56001 and Texas Instruments TMS320C30 processors. In addition, standard C code can also be generated.
- \* Increased limits for model orders in autoregressive (AR), moving average (MA), and autoregressive moving average (ARMA) spectral analysis. In the student edition, the model order is limited to 20.
- \* Includes the macro compiler. Using the macro compiler, a set of PC-DSP commands can be compiled into a macro file which can be executed with the 'Run macro' menu choice. This feature makes it possible for the user to add new functions to PC-DSP by combining the existing functions in a certain way. Also, by using disk file input-output functions, it is possible to interface stand-alone executable programs to PC-DSP, and make them part of an integrated environment. Full documentation on PC-DSP macro language is also included.
- \* Includes the dialog compiler. Using the dialog compiler, interactive data entry forms can be developed as front ends to PC-DSP macros. These data entry forms have the same general characteristics as the data entry forms used in PC-DSP user interface shell. Full documentation on PC-DSP dialog language is also included.

- \* A number of macros are included covering advanced topics such as adaptive filtering and wavelet transforms. Additionally, a macro is provided for importing and exporting sound files in WAV format.
- \* Source code for reading and writing the binary data files used by PC-DSP is provided to aid in the development of child processes (external executable programs that are executed from within a macro). Languages supported are: C, Pascal, Fortran, and QuickBasic.

#### Other Related Products -----

This file contains brief descriptions of related products by DSP Solutions. For ordering information, please see the file 'ORDER.FRM'. For site license information, see the file 'SITELICENSE.TXT'.

#### PC-SIM -----

PC-SIM is a continuous- and discrete-time simulator that is used for time-domain simulation of systems described by block diagrams. It was designed to be a flexible and open-ended tool to allow simulation of a broad range of systems encountered in communications, signal processing, and controls. The block diagram under consideration may contain linear and nonlinear components. Continuous- and discrete-time components can be mixed within the same block diagram through the use of impulse samplers coupled with continuous-to-discrete and discrete-to-continuous converters, and zero- and first-order hold devices. Multiple samplers can be used with different sampling rates and/or non-synchronized timing.

PC-SIM is fully compatible with PC-DSP in terms of the data file formats used. For example, a sequence generated in PC-DSP can be used as input to a system to be simulated using PC-SIM. The output signal obtained from this simulation can be brought into PC-DSP for FFT analysis. A filter (analog or digital) designed using PC-DSP can be used as one block in a block diagram to be simulated using PC-SIM.

A block diagram is described to the program using a very simple script language. Simulation models for a wide variety of linear and nonlinear component types have been developed for use in systems to be simulated. Following is a partial list of available component types:

|                                     |                        |
|-------------------------------------|------------------------|
| Input port (ASCII and PC-DSP files) | Sin(x)                 |
| Recorder ( " " " " )                | Cos(x)                 |
| Adder                               | Tan(x)                 |
| Multiplier                          | ArcSin(x)              |
| Amplifier                           | ArcCos(x)              |
| Dead zone                           | ArcTan(x)              |
| Comparator                          | Sinh(x)                |
| Comparator w/ dead zone             | Cosh(x)                |
| Comparator w/ hysteresis            | Tanh(x)                |
| Limiter                             | ArcSinh(x)             |
| Limiter w/ dead zone                | ArcCosh(x)             |
| Limiter w/ hysteresis               | ArcTanh(x)             |
| Compressor (u-law and A-law)        | Exp(x)                 |
| Expander ( " " " " )                | Log(x)                 |
| Full-wave rectifier                 | Log10(x)               |
| Half-wave rectifier                 | Square                 |
| Zero-order hold                     | Square-root            |
| First-order hold                    | Reciprocate            |
| Sampler (Impulse sampling)          | Minimum                |
| Continuous-to-discrete              | Maximum                |
| Discrete-to-continuous              | Sine Oscillator        |
| Continuous linear system { G(s) }   | Cosine Oscillator      |
| Discrete linear system { H(z) }     | Pulse Oscillator       |
| Integrator                          | Triangular Oscillator  |
| Differentiator                      | Unit-Step Generator    |
| Shifter                             | Unit-Impulse Generator |
| Quantizer                           | Gaussian Noise         |
| Multiplexer                         | Uniform Noise          |
| Demultiplexer                       | Volt. Cont. Oscillator |
| FIR digital filter (from PC-DSP)    | Down-Sampler           |
| IIR digital filter (from PC-DSP)    | Up-Sampler             |
| Analog filter (from PC-DSP)         | Adaptive LMS filter    |

In addition, well-commented C language source code can be automatically generated for the block diagram described. This allows the development of stand-alone executables to simulate particular systems, and can also be used for porting a system to run on a DSP processor.

#### PC-DSP FOR WINDOWS

-----

Has the same features as in the PC-DSP professional edition, except it works within the Microsoft Windows environment (version 3.1 or higher). Tightly integrates with PC-SIM for Windows, but can be used without it as well. Available April 1994.

#### PC-SIM FOR WINDOWS

-----

Has the same features as in the DOS version of PC-SIM described above, except it works within the Microsoft Windows environment (version 3.1 or higher). It also allows interactive drawing of the block diagram on the screen. Once the block diagram is drawn, the system description file is automatically generated, and the simulation is run. Tightly integrates with PC-DSP for Windows, but can be used without it as well. Available April 1994.

#### DSP LIBRARY

-----

Extensive C language library of signal processing functions in source code format. Functions cover a broad range of operations including binary file operations, waveform synthesis, elementary signal operations, signal statistics, convolution, correlation, FFT analysis, filter design and implementation. Most of these functions were used in PC-DSP.

Royalty-free right is granted for the distribution of straight application programs (in executable format) developed using these functions. Contact DSP Solutions for a list of functions and pricing information.

#### ORDER FORM

DSP Solutions  
P.O.Box 341  
Glen Carbon, IL 62034

e-mail: [pcdsp@minuet.siu.edu](mailto:pcdsp@minuet.siu.edu)  
[DspSol@aol.com](mailto:DspSol@aol.com)

| PRODUCT                                                                          | QUANTITY | UNIT PRICE | AMOUNT      |
|----------------------------------------------------------------------------------|----------|------------|-------------|
| PC-DSP Professional Edition                                                      | _____    | \$ 99.00   | \$_____     |
| PC-DSP Professional Editon (w/ student ID)                                       | _____    | \$ 59.00   | \$_____ (*) |
| PC-DSP for Windows (Available April 1994)                                        | _____    | \$ 99.00   | \$_____     |
| PC-SIM for DOS                                                                   | _____    | \$ 99.00   | \$_____     |
| PC-SIM for DOS (w/ student ID)                                                   | _____    | \$ 59.00   | \$_____ (*) |
| PC-SIM for Windows (Available April 1994)                                        | _____    | \$129.00   | \$_____     |
| Bundle-1<br>PC-DSP Prof. Ed. &<br>PC_SIM for DOS                                 | _____    | \$159.00   | \$_____     |
| Bundle-2<br>PC-DSP for Windows &<br>PC_SIM for Windows<br>(Available April 1994) | 1        | \$199.00   | \$_____     |
|                                                                                  |          | Subtotal   | \$ _____    |

Illinois residents must add 6.25 % sales tax \$ \_\_\_\_\_

Disk format [3.5"] 5.25" Shipping \$ 10.00  
(circle one)

TOTAL \$ (\*\*)

(\*) Please include a photocopy of your student ID.  
(\*\*) Personal checks, Visa, MasterCard.  
Purchase orders accepted from academic institutions.

---

Name: \_\_\_\_\_ Company: \_\_\_\_\_

Address: \_\_\_\_\_

City: \_\_\_\_\_ State: \_\_\_\_\_ Zip : \_\_\_\_\_

---

CREDIT CARD INFORMATION Circle one: Visa [MasterCard]

Name on card: \_\_\_\_\_ Account Number: \_\_\_\_\_

Expiration date: \_\_\_\_\_ Signature: \_\_\_\_\_

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

#### 44. Re: Various comments on the newqpsk

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 12:06:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30419971@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain  
>From: Pawel Jalocha

>We should take few our latest talks and send them to HFSIG  
>- who will do that ? :-)

Well, now I did it. Blame me, if you find your texts published ;-)

>> Do you think that 50 mV could get distorted in the (FT757GX)  
>> mic preamp?  
>

>I looked at the diagram and there are two transistors with a fancy  
>loopback before it comes to the potentiometer.  
>In principle 50 mV could be distorted there already... not much,  
>but I think a bipolar transistor does show some non-linearity  
>at the 50 mV level.

Maybe Joni and I should try even lower input levels? I think that would at least sort that alternative out.

>The MIC socket: On my diagram, there is no AF output on the MIC socket...  
>The audio-in on the read is actually the same as on the MIC socket.

This is what I remembered, too. I have now found the extra mic plug I had so I will solder it in during this week. I will solder the up and down switches to this plug so that we can test the rig autotuning. This will be very interesting!

>I measured my transmitter passband today... I connected a simple  
>RF-voltmeter to the TRX output. Here are the output amplitudes:

I plotted this, too. Did you have the mic connected when testing? As the mic is in parallel with the AFSK input, it influences the response very much. The output seems lowpass filtered, and by far not as nice as Jonis' TX. The

excel  
sheet is in the end of this message as uue. Joni's plot is in the same file  
so you  
can compare the results.

>The more I think on the problem the more I am convinced that this is  
>a REAL problem in all type of multi-tone modems (including picollo  
systems).  
>I am pretty sure that this is the problem seen now but Joni and Timo.

>Here are two issues:  
>1. Same as for auto-FEC/Interleave setting: the stations needs IDs  
> known to the TNC and every station should be treated separately.  
>2. The modem doesn't "see" what on the air directly but through the  
> receiver filters, thus the measurement of the carrier's amplitudes  
> will be seriously affected.

Well, how about adding a learn code in the beginning of every transmission,  
every station could then retune the receiver for each received packet. There

would be no need for TX recognition as it would be done every time.

>I like more the idea to measure my transmitter and hardwire the correction.  
>This should fix the thing once and for all.

This works well with one rig, but what if there is no common characteristic  
for the different transmitters, as it seems? A software that would take  
the  
results from TXtest and convert them to weights for the different tones in  
NEWQPSK.DAT? That might be a simple solution for the user.

Here is the new XLS file.

73, Timo

```
begin 644 plot.zip
M4$!#!!0` `` `(`+E;`!`_:@(L:&!<` `` `L` `` `* `` `` `5%A415-4+EA,4^U="W@5
MU;7>> \ [[G.2<`WGQD&1((+R2D-=)@HH)$2(JFJ`H\0(BP@%2DYR0!%'$$FW0
M]E,115*A/O<-5117;5HL;7Q42K2>WQO<J%]:K]VD]1\8I5R%UK[YD]CW/F
MD)!$J6`X]LQD_7NM_5I[[;775!S>V37@X`/;AKQ'3,<9Q$:.=7J(4T>C2F)'
MD!!)^?M89V>G2NX\=?Q+'4<A'5.271E+'',7'V+B@>2%Y(.4`"D1DA]2`) (#
MT@!(`R$E04J&E` (I%5(:I$&0!D,:`FDHI-,@#8.4#BD#D@QI.*1,I=P1<!T)
M`0_2*$BC(8V1-!;2.$@YD` (A94$:#RD?4@&D0DA%D(H5,:>.KA\7D0C\X6Q
MF$(:X=I,KB7=.5)!"U19:`^";HG1=W"X2I_WIL0?_6+"TI>H#>[;%%HE*_\J
M<J*"%ZR00CU=X7%NN]TU: :#>4WD"923:XD/R+=/082B:(-M" FVKZM\ERM7
M:[F8+(7R&\@U0?GPB@LA!HU,THKJ8/[QCAR1I]`^`[,AV;.(L.FGYDS_/2B,
MOLW+==<\]=]%.7U`WOSG2$EG8*D^Y9GZX7@[EY<LS(\U7M2P.AUN9=U!9MW!A
MB*T)+$?>Q8CDA;KH1QP#C<7R#ZZZ_.0JQ<`'[_#3<:->N:_$3?3\H'VGYE\
M3<)UIX9P`V1T)I]_4^&*:]CL3+XV-<$5RVW/Y&04$YE\G7HMDY>_$Z[GP/6=
MX;S]-F8;&M*K!CX_C6(IMIU!7.F*"5%6.X^DWE$))QN69J>)["I1B2$)TG[2
MQ&Q<.J./HIV=2/<Q7D[7]&A2U;S6.7!-3K6C="KUQ[^E=0^1>Y<JM9-I>CK
MAE;OGY(V`IR#`HYD"5=A0$N"VF=:Y)58SP]S#B>'R'[6@YDF*3E*[R3!758,
M+@K4G*BR*P57I2579117K>"JM>2JC>)J%R-EER-45SM@JO=DJL]BFN#X-I@
MR;4ABFNKX-IJR;4UBJH#<'58<G5$<>TCZI*!&N4+XI#/#@3-H?S"!Z(_BH)9&
M!H<02>(S^%0UVEP00149@60;B!D*,ZF`PS4U.@KQ60+SN(>C@]Z!8&(#&(
M`F#Z>9`&NA9Y1JUJ.:2^==ZBL`QUBUS5`ZR--PX_UIY5-"IY%>KP)W2=%;Q
M3>Q^HCF+>B`*9;J)=0$;$YP00W@L*8P9Y0?%9,HK5`*O_S'U?["3<00<I':
M^FJ_QPW5<LCD5H2W7,D@1<J>E`*U==5@?;)K?L+$M[/S`QGY;U&P<9@8?PX
M6*Y=BR"0#3T$YB=AKZ;A6:*LGZ5@4%1KZ\[A=,WZ#M9W&V&,4*H->@FO'?B'
M`V030\,4V9Y"CD#1W!"L/Q, !Y,`W4X#F$`29V!V`RCJ'`"-)4^A=4'NW5R
MT(:=6<548%(0C>8?H<AB&.1/0)=>A!KHAW<8)N!]Q_`,Q%V",,8A_E`#EEI
M#%4*PIPT9LZ,X-@8,F(?&=2LCHTP#A` "'N5Q-$Z6ZIBL;#!0'5TF=<2)(4U=
MCBIG(ZK*Z8M8B(Q@?%`#ZQY(/Y-B%9&FK%&QBL`Y2&=B"9(H(02ID"$V5F57
MT$0:VMI4+Q=FX_000R]63^1(#B9'=8C:B8/8B&8&08(:("6#])2UZ#R;^I
M4_,9M7L+&.$U`K_3T!*L!>H1'D<[B:BW&@F@K-X]M3`[%003"7U^G>60`HIC
M89`^5B8FLBR<+!OL0W+P\FJ.;%QLR+`P+!;.22@[#R`Z!>OAF.;EA975
M_BM[P[S866$],2]RD:5YX;5RL%IA$.A5)NXWD!XFV"N#TK@_`%?TJV"7Z)VQ
M.,P<0.C#6$<E\`]@OP??.MC[2WR`Q.(#>)[(_+`KR! /C3QM2IOB]M5(%PVYC
MYTQV]K/F[&7W]3!=W\#Q7[F&#U*!8U)SW;QZ3J='WRY%TR6+`#:+`+`E;[2D
M6Y4;34^WGP8Z,S@K)RLK?[:<V7!5YAFS<W5_I=N`0T^GZ_%9X475#%ER@1E
MRV"40'R3`$$(MX`C1IARJ67ILMZ)L15)LS-'20KBIF;'S+-%,R3F9NI^[MB
M;B[RCS'PSVVR>Y617Z$($0$WXY\,GN`DP2\JIA>A)RI2RLO-]9@,L2&3'-'5
M]$2='U]#A(<7)[LE,D.69ED0*,<=OAX-*F+/-(#HE%B["L`C`3D.V*DC8E!
MRXE!&QN#YHFB<9LANXWYC+2UDIT$VVR=>!W0YF37@6UV=DUJ<[$K!*#8-;G-
MT;D*HING,0?,CJ0$*@ATUJ0SAZEFWK)P_2JPOWZ.,Q(GV)L6(<&9IQ&*D7!,
M1P@AX?&:X02)`R2KQ%*D?`!A@48H0\)?`C7"A%5@!`?^KB.W[&*$@`RE0%LH
M14CI",DHA4AYI41`*4+*`Z4Z2C%2GBS344)(>72"CE*":QY?&X)L;?)!067,
M'>0`] -6,! (P%HY5,./,"\"_P?@#3^2`8H6\W?_`G`ZZL*9_+Z&,9?1R[OY'=
```

MM[ \$ ] C ) & XG @ 0 - [ V 5 + 4 T + Z 0 V 0 X U 5 [ E ] ; ( A 9 2 0 D 5 6 T L 9 7 \$ 6 " , O ( > @ . 7 \$ ^ X J W & )  
M % / 6 5 9 1 ( 7 L \* W 0 - @ H M \ 3 5 " J K - # 3 U = Y T " : ) & C E Q 8 3 > [ T \* : 3 = # R ' B / T \* 3 \_ 2 ' : >  
M : 3 ? ; I 0 % , 7 @ J H K \_ ' Q 7 H 9 5 # R L \* \ ? & F F R 0 @ 7 ' L + G B M N Q \_ / / [ \ S 3 ] ? B > < 5 = > & [ Z  
M ! 9 X 7 K < < Y ? X G G : ? > \_ R \_ / ? C > < - & Q 0 4 ' G E < \_ R ' @ ? J @ @ 0 N U ) I [ # ] P = ) K 0 T < % R ) 5 8 N  
M K 6 B # G ' 0 - . \_ / @ R N U \ X T \_ 1 C D D V 8 P 2 I - . , = \* \$ 1 4 0 A ' / 3 3 < J . ; ' < @ : Q C ' N ' U 0 ; E 7  
M \ R V < \$ V + ^ - K 9 \_ > > > > > 4 ' , \_ ( K W & 1 1 F ! Y U \$ ( K D [ ? 8 4 0 . \$ N 8 ' O X 2 ' H A F 8 : Q C D G )  
M A / - ) P ! = \ 9 L Z ; 5 Y R # ? ) " Q 4 ^ ' C L , T @ U D ; L 1 M @ . < , 9 0 C A W ] Y M / Q # ' 8 8 8 8 > ) V V F \$  
M G 0 ! G \_ ^ / < F @ \$ 7 - F Z \$ 7 8 ! / . K S - > < V W L ] A M Q \ V F X 1 [ C + # ' 5 # 6 0 \$ ? 8 ' 0 / ? " Z F U \_  
M ^ 2 : / P 3 X C [ ' - X V - R / I N V ? P > \$ \$ ( Y Q @ J E J B \$ 4 X \$ , & R Q ? M 7 N 0 H 8 [ # ? " ? H " 3 ' N S X  
M W 6 < Y ' \ X 8 X 8 " I 8 5 2 % [ 5 ( 2 F P 0 ! J - Z " 2 K D U ( A ? G E , @ S D < 0 . % & # 2 1 ? = / / ^ ! Y K Q R Y  
M ) . 2 R , P 4 8 ' ; " = \* < \ 0 D N X E T R \ 6 F + & 0 ' 0 , K G ( , K " \* V D # < 3 ! Q : ^ S - Y 2 ] K 0 5 3 [ T /  
M 8 N S \$ = D P 1 P V & ; H 1 0 [ U P < - 1 G W 8 = D ; - X < 0 G ] S ' 8 8 8 0 = ) F Z G \$ 4 9 ] " ( R M N 0 6 Z V W < S  
M V & 6 \$ 4 1 \ J 5 X S [ ) < G = P V " W \$ 7 : ; A ' N , L , = 4 - : \ 1 1 G U X ; > : ! Y H < W 0 < Y G Q ' V L 0 = Q  
M Z F , - , \_ ] < # 4 X P 5 2 W ] " \* , ^ Y ( X > , K I E Z " \$ & ^ X T P Z L / , K \_ A Z 1 6 ? G ? \ ; @ @ ! \$ . F ! I &  
M 5 7 @ / E # L = K . N 9 \ # S C & M B ) X - Z ) L K V 3 Q / 9 . 0 P B & 9 G # 0 E \$ ( 2 8 N Z = ; K J F V B \_ U S M Y I  
M " . E 9 : & ; > A F J U ( / 0 # ! 6 A & 6 [ N > B \ T 4 P = 8 \* 1 C > " R ' T \ Z P I - ) , # U 7 D \ ( ^ \ 9 \ ) )  
M \$ 9 I Y " < ; A : Y 7 0 3 % ; 2 A & 8 6 0 4 + G Y / = P C = ! X H 9 E A % D 5 4 - V . , < B 3 ( S K 5 6 ) ] ' 9 [ J F  
M P T N \$ # G . + W 7 L Z G " / Q . ( . K W 5 X < 8 ? 0 > . ' % W M 3 ; = ! P # 8 H L 6 Q = X \ & , [ J ; N Q \ L Q J 6  
M N / H \ " ( F : G N # B F N Z ) J ^ D I \ 8 I 8 4 - F 7 > I X 6 1 - . [ [ C N \* 1 , : \* ' = X @ \ = C A ? = Y X L < , S  
M J ' 7 L \$ - \ [ 2 U B A \ W A E J 9 ( 8 T M 8 K F N 4 P = . 1 9 Q 7 D R \_ 6 1 1 : - K Q A ? G E 8 9 \$ S ^ 0 > / < @  
M - 9 ! H 9 X N A " Q 9 # - S R , 3 X - % T \_ 7 ; ( O G % ' - 0 J / ; \_ = G . 0 + ( 9 : \$ 2 > ^ & , I % K Y \_ X 8 K B Y  
M S P Q ) - 6 ! 5 T + & + 8 # % , \ T 0 0 A M M M ? # ' , \ 7 9 U , 8 Q G " , ; U Z 5 \* Y " > ; - Q X H ! & = Y W ! N 1 "  
M - 5 < @ [ I / 5 0 / R 0 ' F - X @ > 5 \ N - X " " ] E J R 6 I , + Z / \ - ; 9 8 I @ C I ! 5 : F R & 2 ) Y / J Z Q K # <  
M 5 ! ] I C 6 ^ 3 - H S ^ ^ [ , @ ' 0 \* \ > " E C & / 2 R 3 7 M 3 H [ . < = # X . = D 0 ^ [ G M " \ 6 ) & 2 ? # " W V , C  
M 3 J J 8 D 9 U " R Y K , Y C " R R ! @ S > C T A P % Z D T & ) & \ W 0 Q H P \ L \ J P \ \_ ] 0 8 T 9 ( D W \ [ 1 K - 0  
M 5 F - & N G Q N - 6 ; \$ : > ^ - @ 9 B 1 \ W N / & : 7 J 8 D 9 . \ ' . P U U / ) B ! > \_ 0 ? G P K & \ % L P ) ' \$ \ E D  
M V P Z \* R 0 ' Q 8 B ? N ' E % M 4 T G ) F 2 ? M & Y X \$ \* ( " \$ @ \_ 4 ( ' F [ / 8 W ! 4 L # F 3 R 3 3 Z \ Z 6 = ) S P  
M G ) Z T [ ! G X ^ : 1 = ^ < 9 I 5 S D Y ; ^ = X \_ L X Y Y [ \_ A \ C / 8 8 8 3 A , 1 S Q [ G F H : ^ + 7 Q V " G \$ 7 8 "  
M 0 " Q W , ; > A > A # 7 4 8 8 B U E X R 9 0 ; M 0 \_ U R M ( / ! ; B / L : K C A ( ; K = , # . 8 ( \ 1 ) @ # \ A \ + 7  
M M N 4 U < M A K A + W X ; + # > T N N ? ? - [ & 8 ) \ 1 ] @ % \ T + = V \ X W 3 ) 0 8 G & . \$ \$ @ ! ^ & 9 N U Y B # ( X  
M T 0 C C L \ # = = \ P = 0 8 . > P W P G X 3 = \ ( ! P ! V O C U X ^ M Z A Q \ H 1 I B J L Q 0 ! 2 # # \ U \ \$ ;  
M C Q [ 6 K L - X 7 5 X [ N ] E Y , = , Z , A ; 0 / [ Z 3 W 8 \_ " ^ S ; 4 3 + J ) G 6 0 9 ^ 8 4 U # ! W ) X H ^ Z ^ Y K 5  
M 3 ' / - T : , ) . : 7 R I 4 Y X 0 9 = ' C V ; ] [ ; / \_ 6 ' # \$ 6 \ ' > ; > B \$ 4 % S R T T , E 5 ) 0 5 , U 2 4 W Y 3 1  
M < O M U W @ I = J \$ @ 2 \$ 2 ? 4 2 " 8 & 0 X Z 6 8 C " D E # 5 G 4 - F [ 1 5 T 5 N I " 2 I \ L I 0 ? P \_ E U S [ T 5 % 7  
M A 2 Z D ) . E " 2 M Z B M X ) 2 U - H K = " \$ E 2 1 = 2 R M Y ] P \_ N 1 % X \_ I 0 T J 2 + J 3 T Y Z 0 = S D W ? : 4 /  
M \* 4 F Z D % + 5 \ N " Z ] = ] \ H 0 \ I 2 ; J 0 T I - 0 7 [ \ W = ( 0 ) 2 \$ E 2 1 = 2 V 0 / 4 D K 0 ; M W \ N # R E )  
M N I # 2 U > D 5 # Z ^ > \ 0 = R 7 4 A ) T H 6 4 ' E L Q > - K 7 5 @ \_ U ( 2 5 ) % U ( Z ) V W ? ^ L ; - ^ M # 2 I ( N  
M I \* 3 G # A C A @ " F < 1 5 4 8 0 T I - 2 D C ) + 2 T G 6 D A ) ? 1 5 0 ( K L A [ : 6 E 6 8 2 4 ; L V 0 ] F \_ T ? W 4 >  
M W N ? = V , / \ 2 [ 2 L \ ? Q % ] ( K U K ) Y " ] Z " ' ^ Y \$ 2 2 > C J [ ' W P > B Y \ P K ' , X ' [ @ 9 \ N N D 2  
M Y 6 F [ D K 3 \ < 3 Q M J K 2 J & Y Y V ' 0 3 G . X ' W ' \_ U 2 K ] ' 3 G @ [ X . 8 ! M ! " ] G = + \ W 7 ; J > R \ T  
M ! ? ) / A \_ 9 = " Y I 0 " ; 3 ' 8 ) L - \$ / T ! 8 8 Z \$ 5 \_ N T F 0 9 7 F H ! = V E S E W W 6 6 0 = : # ^ X > S ) /  
M . & " D > X W W 1 / S 0 M Y S : # [ 8 # : S U J F 6 5 L O G 0 U R ^ ^ ) S N [ P / @ 6 H ) / M K H L D - Q N 0 " T # :  
M Z 2 2 6 7 3 A 8 4 . T 0 3 N H 5 N \ + . W & [ < / ? . " 9 9 V ( 0 C \_ 8 M J % , W 3 5 0 " M V 8 : ; 1 + @ 2 / 9 Q > "  
M 7 : 8 + E 8 I = \* / 9 % V X 5 % ; L @ ) Q S / + G S \_ H ; S 0 ) @ B & # F 2 \_ 8 D \$ > E F ) ; D / U Q + , B ; ? 6 ! !  
M N B > S Z V & X % Y 4 Y G ] U K < U Y F 0 5 \ S 4 > Q > Q ? < % ; / ? J 5 - X ) < < ( \_ A ^ - \$ = Z ^ : 0 ? M C [ L F U  
M < V 6 0 [ % F ] [ 8 ' [ 5 > U + " W 5 0 K ^ L = ; 6 ^ 0 = J = 5 N W ^ ? = [ \* U F U B \_ Y = & + [ 7 Y Y ? / ] L ] W / Y  
M \_ ; / = 6 P K Z 9 [ L ? \* > R ? [ ; Z W J ' ^ V ^ Z [ B \_ M G N V T + ] L ] V K 2 0 I G N Z \ 0 [ 9 \_ M 7 E K 6 / ] 0 =  
M \_ , % \_ M C M \ ^ @ \ W : = ^ G ; / 0 \_ ^ 7 M & \_ ; L : / X - ^ : 3 R \* > L [ ] 3 0 B / G 5 I E Q = I F \_ 5 . = 6 M  
M 7 ' W \* U : 0 [ % M U \* 0 0 H - . P ^ H < + X \$ = C T \$ < 0 A 7 Z ' Y B \_ ' K = I ^ ! > 1 2 K R F K ] U Q R C 8 " ) N #  
M 5 ) . I ) \$ P Q ' 8 ' [ ( ^ Q Q ) = M V + ! # . F I M U 5 + Y E ] % 3 ( : \_ Q . W ] ) ^ 6 H Z 2 4 ) Y \_ . Z ( [ J Y :  
M H - 6 B C \ @ U U \ K @ W \* " R Y ! O I " T 5 7 U P A 5 9 A ^ 1 D 3 = . U \_ = J , G < + W @ 0 5 F @ ! . V ^ H ' 4 (<  
M 6 H > @ ; 1 & T + 8 \* V 5 M # 6 " M I R 0 5 L N : / , \$ ; 9 Z @ 3 1 6 T J 8 ) V 1 - < V W D ? ) T K > " = E B Y 0 ] I \  
M 0 9 L 0 : , L \$ ; 9 F @ W 2 ) H M P C : ? 8 ) V G Z ' ] + 6 A / " U J ' H ' 4 ( V E Y ! V R M H G P C : ) X ( F 4 9 4 F  
M B 7 : D " E J J H ( T 1 M # & - E ' 0 ) @ K : = \$ & ; + F A A 0 0 0 3 [ L P Y K V ' . Y " K : J \_ = ) P = V I Q S  
M 5 \$ > W Y G 1 W Y " \_ H 7 - F ] : \_ E < Y M ! % 7 0 . ^ 1 \* 5 F 9 5 @ F - V X 0 I E M B M % 2 4 & 8 I 1 C ) < + 1 N 9  
M ' 9 I U & ? D I N 9 5 < ! + J X 2 F ! X O Q A H # 8 ' U 0 Y Y E D % ? % \ ' X % N 1 U > Y [ D 3 < O P < T + 0 8 - Z X C  
M ; 2 Y V ? Q N D - 6 0 M ( & 0 > O C K ' D C W 0 4 I E > 3 5 L ? B / P / T 3 N ( [ ] ' W D V 0 Z U ? D 7 K ( % E M : M  
M 0 ' T : J \$ & 6 U \ G N G R 6 / D G ^ ' Z W ; ' 7 @ 7 . # L B K X G C \_ ! ^ ! \_ ! T K [ , \ A \ E Z P & W = ; J B 0 < '  
M ( ' T \ V - ^ @ 0 \$ ^ @ 0 % . \_ O W + J ] U < J 3 \_ W ^ R O ? V ^ R L J U S Y + K G U 1 7 \* F " \* Q 5 T , 1 9 7 \$ T V -  
M H 6 \$ K % : X T N ( O F 2 F , : M 3 \* \* : Y ? @ V F 7 ) M 2 N \* \* X . J 7 ! D Q : L B Y 5 \$ 3 C B @ B N B " 5 7 A \* H S  
M K Y , Y ) O I U Q K R ^ Z . = C K ' F < ! O V K 2 M N Q 8 P < D S F = 7 < B . - 2 \ \ [ \ X S F : P H U S % 8 U G W Y .  
M 9 H D Y & 7 N T + J # : ; - 7 F 9 ) : 8 D U 9 < E 5 % < M 8 \* K U I \* K - H J K 4 7 ^ U 6 G ( U 1 G U " Z Y V 2 Z [ V  
M \* \* X - @ F N # ) = > & \* \* Z M @ F N K ) = ? 6 \* \* X . P = 5 A R = 4 1 Q ; 5 / < . V S Y - H 7 Q 9 4 F 9 M ? ^ F + . K  
M @ 5 W W - \ ^ N - # & [ ] L > < 7 9 5 + / + 0 2 Q . S : ' W - V < 2 [ S [ \$ H 3 L V M \_ S - G % N 5 0 D ] G R R Z V : #  
M - D . , Z V ^ 0 [ 6 , ^ 5 ? 8 Q V R 7 C / N : L R 5 R 2 P [ 3 R Z G T R \_ ( W 1 0 J \ - O E ; ) [ C X U 5 P F ^ \$ L R  
M > R ^ . S U 7 ^ . ) Y + < B J 2 D & ; > L < 6 > H ] J N 3 ^ V ; % " ( I X Y ^ B U E 3 C F , L \$ Q B Y ) # ) < < ^ V /  
M P 9 4 \$ ? ) ) ^ 5 \* Y D ] @ \$ R < B 5 ; E ) 5 ! D H G - Q & 6 # & J @ V 7 [ 8 H R R 8 0 M 6 \_ T 0 6 R T 8 U H \_ # X 8 K  
M 9 0 \_ B ! T % ] < D @ I 2 . 7 / > ( ' D ^ 5 ) " @ 4 R 1 2 " 3 ' ; F % = U Q 4 9 ) 9 ^ \_ V 0 1 0 # = 5 & 2 ) 0 ) H '  
M : 2 < A @ = P L D ) L ^ V 4 ! \* ! \* 0 0 . X ! 9 " L I < @ S ^ G D : D ^ Y 2 ) I ^ W ! ? ( & / O ( ! ( ' 8 X % L @  
M 1 5 9 ^ 6 0 D : - 0 G = 5 2 ! ! Q F " , : ] = @ % 1 " / \$ U / F ? ( ; D @ 9 L \* J > K 4 , J \* 1 M ? & ( < ( ) , !  
M X ; J ? S , 9 % U 7 T J 2 9 D 7 7 ] 0 2 & F X @ K G A [ ( 4 ? 4 W ( I > J X Q S R Q P C ^ , G 9 : B 0 F \_ A S \ M 3 ( '  
M / : ; Y H I ^ # \* 8 H 8 9 + . V B % ? 2 ( 2 K G ^ H H 7 \_ G E 8 \ K # 7 + 3 ' G - N 9 ; E X ; R & 5 6 C 4 ^ + E Y 6 &  
M % + J J 6 U . 9 I 2 G ( S \ \_ W \* \$ B 2 H E 0 9 ; + : J ; Z - [ " > X \* L < 1 \$ R . - 2 Y \$ X A S [ ' R 0 C F H E 0 Y I  
M W A 1 X D 8 ; 3 B T S 7 2 Z Q " I 7 J 8 K > 9 K . ( H \_ ; N L X > 5 B Z V U L 7 = J M 1 K . @ G T @ U K " @ ) \* > @  
M L # A 4 2 % \$ D & M L Q \_ \_ 9 2 J [ " T K ( R R U R C : 3 Q 9 @ Y 2 Y D \ U ^ @ 5 G W = 3 W A ] ; , K ^ 7 D : H \$ A =  
M H M 0 P / R < 4 ! % ) # 1 I D - M A 0 # 7 . \ Q \_ < K / \* 2 D ( % 9 3 % K - \ 2 I 7 Y 6 D H S Z & 3 M B Y R 6 Q / # : ?  
M + B I H W # 6 A W 4 . : H ? ; / Z ' S I ' I B 5 Q X L I F / = W W 8 E ( = D T V 1 A Y Z 7 W 9 / X Z < ] E Z \_ & C ' 6  
M \ 7 8 W B , 0 = S A \* ' H , Q \ ' + B & ' C O H L ^ 0 ! / H " " = ! ? D P ^ T 9 9 ( , \* 1 6 2 F @ ? 0 ! P \$ V % / C ,  
M H \ 1 & ? R Q \$ ? 1 1 B ( 7 = V Z U : J A & A 1 \* ) & ; O H F X M 0 5 ^ < 9 > ] G 2 K ' 9 Y N E # , . R D D D K 7 \$ \

```

MHF(V%Y,@VH00N\;_L4QNG;E=X+;3Q_YV*A:;SU]'#$0NZ.46:N07B>CKB<OW
M]K+&Q2Y'/SY>-CZ'P#]A0UA7C'%60L[M`8_*SF@Y(1Y)=2FY7U%SA[3<CZBY
M0^;<MZBY2[3<S6KN$G/NR]3<I5KN"C5WJ9;[N^BG[W(<@N(92KNMW:;X1T'5
M[]%6'_@_KD31ZC0!9S$^!NB<IG4:(ZLWT5-(Z+JDPU&-)KRBZU/;Z?:AH3L_K
M&=>7V$0]%. (2^J%?IK#)<7HI[[6WZ#AF9_93_-,*[87:N'KQOUT*^@T?8L
MB_Z)69&IRSW=M!7=J<.)V(K>:*~12Y2%UJB1@C\BI9L@7T"4G#)-3[-]?_+
M]HQ1/E':DV@H[Y#R[$$SUA/5/5^(J1>GM)R[:[H*(0VVA_HD@$>VWVEWH([A\
ME_$A05*WRSA^/^CK&\L[Q_H2W=L-^B>5"5UZCA4K:A7KF9'Z'(!8/@>(Q[4_
MSI_EHU:HK>;/5(WOZ2QM06MP\9#9''[LUG+RE1JN0++$>"EMH9;60'G'I;6
MM[:PGR0@9WWE2E?^3Z&J&7)IJ%0^IU8F2Y&&N_49=0T1^>*ZJY?6SV-RL&[C
M<F&7*B^H)`H-TXR(W!#.D5L7@S?<$@XWM,A+EM:UAN5%DB"/"5/,/K_L!?!?
M-&*V[0-[W[+Z/\!4$!`A0`%````@`N5L<']J`BQH8%P``&P``H````
M``````@````````%185$535"Y83%-02P4&``````$``0`X````0!<````
`
end

```

## 45. I measured my RX

```

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>
Date: Mon, 28 Aug 95 11:18:00 eet
To: "'HFSIG'" <hfsig@tapr.org>
Message-ID: <30418DB6@ntcite01es.ntc.nokia.com>
Rendered body type: text/plain

```

Hello,

I came home last night at 22 o'clock so I did not have time to make the TX test yet. I think I'll do it tonight.

I made the RX test this morning before leaving to the job. I made it in the same manner as Pawel: I switched the marker generator on and tuned the rig around the marking frequency. If I do not recall wrong, the S meter measures the input \_voltage\_ so that one S-unit is 6 dB. In order to find the half power bandwidth one needs to find the edges where the voltage drops 6dB from the passband peak, i.e the power drops 3 dB.

Well, the result was very simple. the -6 dB voltage points were exactly 3,5 kHz apart. There was some ripple in the passband, but it was not more than 1 dB (= the thickness of the needle).

I'll be back with the TX results when I get the tests made.

My rig is an FT757GX MK I

73, Timo

## 46. FW: Various comments on the newqpsk

```

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>
Date: Mon, 28 Aug 95 11:18:00 eet
To: "'HFSIG'" <hfsig@tapr.org>
Message-ID: <30418D39@ntcite01es.ntc.nokia.com>
Rendered body type: text/plain

```

```

From: Pawel Jalocho
To: daniel.brynard; Johan Forrer; Timo Sivula; Frode Weierud; Jarkko Vuori;
Joni
Subject: Re: Various comments on the newqpsk
Date: 25. August 1995 23:39

```

Hello everybody,

```

> First: Should we either move this discussion to the hfsig group, or create
a
> new group
> for DSP56k qpsk testing and reporting? I suggest this, as I am getting fed
> up with this
> carbon copying every time :-)

```

In principle I could create a simple mailing list on home-gate but HFSIG is a better place maybe. We should take few of our latest talks and send them to HFSIG - who will do that ? :-)

```

> How does the headphone output relate to the line output? At least the

```



> headphone  
> output is quite noisy.

I would expect same amount of noise on the line output.  
The "ideal" solution is to use a resistor divider just before  
the microphone input.

> Do you think that 50 mV could get distorted in the  
> mic preamp?

I looked at the diagram and there are two transistors with a fancy  
loopback before it comes to the potentiometer.  
In principle 50 mV could be distorted there already... not much,  
but I think a bipolar transistor does show some non-linearity  
at the 50 mV level.

> This seems like a good idea, the only problem is that as far as I remember  
> Joni  
> reduced the output power until it could not get less, and it still worked  
> ok. The 757  
> has quite good dynamics in what comes to setting the TX output. Remember  
that  
> we are only 10 km apart.

Your antennas are too good - cut them by half :-)

The MIC socket: On my diagram, there is no AF output on the MIC socket...  
The audio-in on the read is actually the same as on the MIC socket.

> Sorry, no. T4 runs on a TheFirmware TNC, and I have to use a KISS emulator  
> (TFKISS) to interface T4 to an non TF TNC.

I always thought software is so much flexible :-)

> Why not make this automatic instead? Let the modem collect the number of  
> errorcorrections and then according to some trig-levels switch automaticly  
> to a less complicated FEC and higher speed if the link permits it, and  
vice  
> versa.  
> You only need to find out how to tell the other modem about the mode  
> change... ;-)  
> but with 15 tones to choose from that should not be too difficult, I hope.  
>  
> Now the modem would autotune and fit the errorcorrecting code according to  
> the  
> band conditions, what say?

In one-to-one QSO this is "easy" but if we want to keep this modem  
compatible with the ideas of packet network the whole issue becomes  
difficult. You are going to send packet to two or three station  
and for every station you may need to use different parameters.  
Thus you need to give every station an ID which is known to the TNC  
(KISS TNC doesn't know anything about callsigns).  
Then, the TNC would need to gather statistics for every ID separately  
and then decide and agree with the other station upon the parameters.  
Phil's new protocol probably does something like this...

> Here came the Problems. I pressed w once, and the hard disk started  
running.  
> And it did not stop, not until I forced the system down with ctrl-alt del.

Definetely a wrong behaviour but why I can't say... maybe Jarkko ?

> What does the S do? What happens when I press w, or what should  
> happen?  
> How does this SPY work?

When you press "S" the PC sends the request (just the character "s")  
to the DSP to send the "spied data". The programmer (that is me) decides  
what are the "spied data" at the moment.  
In response the DSP sends "P" and then 512 16-bit words.  
This words are being displayed in a form of diagram on the screen.  
They can be for example the FFT values, or just the input samples,  
or samples after a filter - anything: it's up to the programmer.

When you press "w" the SPY.EXE should write those 512 words onto an ASCII  
file - that's all. Why this simple thing did not work, I don't really know.

> I will be out of town for the weekend but this test seems to be a good

one.  
> I'll  
> run it when I get back on monday. Could you make the test soft so that one  
> could give the modulating tone directly as a number? Like 100 (enter)  
would  
> give 100 Hz, 3000 (enter) would give 3 kHz? This would be very useful for  
> other applications as well... A settable sinewave generator on the DSP4!

Well, you can make it multi-tone, multi-waveform, etc. :-)  
But for now it has to wait... or there is maybe a volunteer  
to take this job ?

I measured my transmitter passband today... I connected a simple  
RF-voltmeter to the TRX output. Here are the output amplitudes:

| Carrier<br>number | Amplitude<br>[V] |
|-------------------|------------------|
| 1                 | 14.3             |
| 2                 | 14.2             |
| 3                 | 14.1             |
| 4                 | 13.7             |
| 5                 | 13.0             |
| 6                 | 12.1             |
| 7                 | 11.5             |
| 8                 | 11.2             |
| 9                 | 11.3             |
| 10                | 11.1             |
| 11                | 10.7             |
| 12                | 10.5             |
| 13                | 10.6             |
| 14                | 10.5             |
| 15                | 9.7              |

The difference between first and last is about 1.5 which make more  
than 2 in the power level.

The more I think on the problem the more I am convinced that this is  
a REAL problem in all type of multi-tone modems (including piccolo systems).  
I am pretty sure that this is the problem seen now but Joni and Timo.

What shall we do about it ? I can see two "main" solutions:

1. Make the transmission so narrow that it fits into "every" transceiver.  
But then your bit rates will be so low... or you will not be able to take  
those real advantages of multi-tone transmissions. And when you make  
the transmission significantly narrower than the SSB passband you get  
additional problems with the AGC of the receiver, the anti-aliasing  
filters, etc. The whole excitement falls apart...
2. Stay with about 2 kHz bandwidth and let every operator adjust the  
passband  
center and carrier level correction. Thus keep higher bit rate and take  
some advantage of spreading the signal over larger bandwidth.

> Pawel, excuse me for my mad ideas but... Could this be done?  
>  
> Make one of the preamble tones being in the high end of the pass band.  
Like  
> 1500  
> and, or why not using 3 or more tones.  
>  
> The receiving modem, during tune phase, measures the level difference  
> between this  
> tone and a tone in the "flat" response area. The dummy bits (pre/post data  
> or whatever  
> they are called) is then used to send the measurement data in the next  
> transmission  
> back to the other end.

Here are two issues:

1. Same as for auto-FEC/Interleave setting: the stations needs IDs  
known to the TNC and every station should be treated separately.
2. The modem doesn't "see" what on the air directly but through the  
receiver filters, thus the measurement of the carrier's amplitudes  
will be seriously affected.

I like more the idea to measure my transmitter and hardwire the correction.  
This should fix the thing once and for all.

> How about automating this, too like above? Is it possible? How much

> processing power  
> would this correcting need if it would be done "live"? This feature could  
> also help against  
> selective fading in some cases on HF.

This might be indeed very interesting but is selective fading  
"stable" enough? I suspect the "notching point" moves all the time...

Pawel

## 47. Excel plot of Jonis TX

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 11:19:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30418E2E@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

Hello,

I plotted Jonis TX response with Excel, so those of you having Excel 5.0 can view

Jonis TX output response. I plotted it against frequency and calculated the output mean voltage from the measured power assuming a load of 50+j0 Ohm

$$P = U^2 / Z$$
$$U = \sqrt{P / 50},$$

Peak voltage would be  $U = \sqrt{2} * U$ , if somebody wants to plot that, too.

There are 4 plots in the file, voltage vs freq, power vs freq and two dB plots.

The dB vs freq is calculated with the lowest measured level as reference ->

$\text{dB} = 10 \log (P / 4,6)$ . I plotted this in two ways in order to make it as easy as possible to use the results for correcting the modem output. The output seems quite nice, I think. It is within 1,2 dB within the whole passband. This should not affect the quality of transmission. Would my loudspeakers do this from 20 to 20000 Hz  
I would call them dreamware ;-)

I'll be back with pictures of my own rig when I get the rig measured.

Please let me know if you would like the plots in EXCEL4 instead? I do not have LINUX, so this is what I have to offer...

73, Timo

begin 644 plot.zip  
M4\$!#!!0` `` `(`+Q4'! \.;(S.D10` `` `F` `` `\* `` `` `5%A415-4+EA,4^U="W04  
M59J^5=V=?B0A'\*==\$PJ'4((G9!7\$S@JC\C+'9X"0UQ@&(@&LI)T2(\*.R\*S1  
M">HL/E!T)I`116=UF'78#3KNPV5F\,AF1M<Y.'/.+KL>6'!T=? :L1U'9,S@J  
M0?\_W[JW'MT5\L)A38IS4]7?\_[BW;MW\_\_0?\_JZZ^<3+[W%- 'Q[S%; ,<US,4N  
M)?PLS80I>J\$CR)B\_]Z42"0\$G!@^\_E\=7T"YI!>W\_BSQF70Y(V9^\*`\$HZ5'R  
MH&1"&0\$E"XH'2C:4D5!RH(2@A\*'D0LF#,@K\*:"ACH(R%,@[\*>"CY4`KTN@KA  
M7`1E\*I2)4"9!\*88R&4H)E"E02J%H4")0RJ"40YD&I0)\*I:YG^.C?<2.+P[\V  
MZ-\_YK'G.+>QVUI<C%T:!!T(7S0="G\$GZ<DQ>8>>]9MW)-U<Z7%!=<M^M8'=5\_  
M^"00\$8!9R'P\_09'QX3B\*BCGM>JB\_D36S9>QF]F>LK\=(IBHX![KTN:^W<M\_0  
MSQZVDNV\$^A091NK[&^`I:(\$6M1#2QAK@NJD'/27]N/\)4\$X4,3FG#]O/T#T4  
M>/JN`!^[=M0%>7I)PZ:6>&M\2YLV\_UN;8MNU:'F%MB;><DOKMEBLC58'=0U;  
MMD3))Q!'^4JDE\$=N8ZX!" ,6ZS^WY\F//UVV+?C<PSXV=? (+\_X%T.U8!V\*-  
MW">AWUG.^!KD=X7<\_GQ`0Q\VKHC[I@HX8[WSBKB/VE[\$\_51'\$:\_\_\_?C@OA/.\_^  
M0G[\_I,(UYB\8^6^%:S%=3B(GJZ&,=W;^55QI:AH;%B;6\FDLZJH1,E0S[!F  
MFM/R'9^L)!\*(IY,LQPT-AE;!Z\RI/CG7C=H5:#\_`^EP5?8K298IHFT#.,;<-9  
M[P^J\12XA\$(2(16),%[E0.N+'A5ZOGQ:87L/#M#/5ADTQ+1>R<'KB:DD%(`  
MC23572>EZAREZI\*DJ540:-4?9)4DY1J<I1J2I+JD%=(CE=(25\*=4JK34:HS  
M2:I+2G4Y2G4E275+J6Y'J>XDJ=/ ,<M0\$C3(I.(I\*I?C?39V%,+H;S:A@7NP&Z  
MBU4#Q[!@",S/&"6?ZDN[XB`.MA5!G[3RS" `N\+\_)-[4T\*JMJM>^`M\_>MG%K  
M3+NU5500\$MNQ,]:TZ79M<C!-YQ=-X"O/? )HM'LEPJ2/,;6V^=PXBH'4[R7\$  
M-&SG/\$2(RT`>?9@OMX'<MY\0CX'L?HP0TXS5\_> \$>`UD\%?"\*;\*OT^ (WT`6  
M'R0D8&K/DX2D&TCG(4(R3"U\BI!,`WGP:4),=W\_?#PC)HA]00+1Q]KF""D\*"  
M/#1[8?VKHUS(D>PDGI&\$%/\TA>??3B-D)PDGA'AQ>\_?L#Q[\*><)>\$S+YXWTW  
M-#W)D=PDJ;PDS:,(>7/ILJ/\_^5DY(:,)&\_A0<5G5G%D3)+FL81,Z=QV9H^W  
MDI!QA.0W?W\*1Q&.C+?4CGV\GLVB9^!YYV;YF6?N-\*@1=B38;UE]`CF\*LH<  
M&I@^%KGU5.:J5Q/4U>M\*3V7Z??` \ /1J,M>=> \ ("B%L\`-YMW.8[5XN7?(?4=

MJ/<I"EI0#`Y+H\KX<6ZV<:X&7A<N\$6FAK`)G'OY5E2`B:C`HF]7U>J&R[T`W  
MV=4A&)ZHU046A.<L(>XV:)6!`?KTRW878.KOH<R-\SRD#XT224+&505`SFN  
MH&QS5AIH\$`A:~@`W+R@`P?L@B`V=FX0Q\JOH,H:,\*\$/8&C`([3`;/I3@?8C  
MD'T?9\*Z#83V>).P`2FCZ52AZ1<AILET39T&P-(6.U\$>!8I^JFN`Y0`M/]#GT  
M3HY354B/4`&J\MJF\*IP)U\$6[<#IRR>G(7,46%(2YX`2<3T`YKIJ`JBCQ]D9\*J  
M'IQ^E#58@RIKB\$\*(HJ+FNP-!EA[>[L(<X(9[.VWVY;+)Y0(14-)`2(Z<10]  
MT:~@?EDL8E!;)\_Y3DXD\$GS,>VC,8Q+H96K2S\_55X&\_1,B`V#\*S:~J-E7ZQ%  
M2W74@=0[ L?A+UJXRN,BR@K@W^MHSODF^\_&T<6-FZ+=0K\XAV@/TMXC^CB`3  
M>9.N^\$9\_1)CJCOW\;;N\LQM:=BXG>,\*X,?49%QUX'<Y\\*?6?~\@1=ZHW<~JW  
MCX.9:/2\$R(0)%>NTHL9;BJY95V;ZE>\NA&DCWTQ?>V-L\WH;4Q&.G0)"RBML  
M>B20[YX,L]1\$&Y=9FXGU6LBFS-Q05JJ9JME0!CKM"/(4E169?L\_94(;R4RSR  
M&QHW&\*5UQ\$I+W^3\_#Q8\_\V5\K)A9A5F4-<R>[:]'?,@8V338V^\*&33I,;?G  
MG.XQ\_5>AL>;-Q3\$\$(Y=N!RF]A)C7R%,8=:^\L"3`+Z+5FQ\*"BR2`BM-@?F3  
M,1XQ:CZ! [5?=-CN2N`YNSV-SB/;W73.:??2&5),= `ZU>Q] [8!6@D(=,H\_`T  
M\$H%GB@V@`H&#DPV@&H`'2@R@!H\$`IAA`%(\$]I08P`8%0356`6@1V1@Q@!@\*-  
M908P<P],>K%R"NYXPRH063\_-A%0BLKK"A%0ALJ32A%0C,K`A-A@<EVU`8DB  
M\$JTQ(=/1.W`'\$"0`D\*Z\*1^QA)Z`[9F9@PA<GY05XNQE`"%YGTQ`/PISS^8\^  
M^LV2FY?/WD!X\*>%3Z?INNFXW+9`GX2-A/N5^6MUL4>X`CI?>05IF;5`XRF&!  
MTM3&DI99;K:~8<#Q;7T503J`Z^("E,#3)\$Y:I\_\*1U\$V-`&R#F&F0)\*.F"JQ  
MWMMI50B82V);~\@3FEA7/:Z\W&HVZ00#:)4Q\$S`!R6%PQ2>A"DD"5,8\$J;0  
M(TSAIIA"C#"%6\$`>(4/H0I9`A3F!"FT"!X4`80H`P+?NSH&+>.:PR`\*6:  
MX[?%6K!>E>I5YK0#I[\*/\_O+K<W<:UX?^'/6PD)W\*\$%>.[S:HVMWXM\_ENG1/K  
M`4D=DP7K0ZCW5A[ 'I<%BW45K5+\$4] [&M,>4+#U,0/N8.`QFNLEQ9Z2R\_BBW&  
MA,;<\$.-R`LB)8,6`C`E=CI-=K4NYK:2W4`6P0:2/5:RQR:=9B6G`5D\$. \$CV  
M6LE>((N8!,"D\*!`EG4^ZWD0VVI@6LY`"015"\$Y`0K.1W((D)`<H:5G&%K6J:5  
MG`ED\$3LA>825C&C`\*20G&4E9]EN3!%DMYI#1I`%S=M<I[7%M9K(=&T-2KEI  
M`,R]~<D59\_U0548T%:7<-`F`ME-`V`,RP^PQ9#P#]A`Q\@Y:~/G,\*5.:60>  
MK@;'P[HC,UYY\&\_?C5NYKJDJ^%DEZ46-Q\!AG`P]KEE\_(\_)031/98R1Z;  
M=)J5C.,AJW3!`7<!]&LB>ZUD`\_]UNZ=^GY6=(K+/20;9E/NM9+^M:0\$K&<?#  
MB35G6\_[J\`^(G&XEI^N1/G]WX>;CP2!GV)J6:27C>"@K&5/2.08\D4=8R3@>  
MUGSZVT/?=I`1,ZRDK-L-Z8(\BFH=P7,KM?`2XM00<2\$X;="X;<X;<`83V&  
MWUXXATWI#7/X[=YY`C.X&.&WFRH;2/B][X`&31I`^+U&AM]\NAN\+L!:+4P  
M\Z2!~/LGM0`[`LWY,=#?`YF95T7X\_1(\A\_2P^!)5R;\W@KE+%3SSW`. \*SV%  
MW^,=JL`^GAMS%=%1)[X8<)X;7<T97^)`V\F57(SV`H<2GA\D6"RW>)C#(7:  
M;7[UDD[M6D\CS!\_,2!5^8]9NT>^4?7Y=SSXGX+G=X3BF50>0?49\K,P^+<E  
MDR7WW`/JN8L-IYY3IY\_]3H>(5/JN9J.L"GU;.:ALQ/W\_B%3ZMG,PRVJ`P`>  
M.:;4LSYSF%/9JF\),VC3.W),:6>Q=K#2#V;-8\U2>694L^~/7FFU+.H7:2>  
M7>3[/./3[.J&%&CW38)\$\*M\_7?/NIS\*^\_~BB^+U/OC0[Z/JUZ@\*GGR"]`\_&5  
MWN#YOHC\*4`\_`'ZIIY^/IETL]#Z:\_R^YB;F25=%G%874=NL,:)\_-IGN0>((:  
M/62&EWM(?X\>,MQ3%90KKJ1\_S`OB+/>]70K(@7J15`[E+I7[E2<`/F568JS  
M7\%<3XGT\*RVQUN9X4VM,:VC2(\$+2UE96:-OC6TN63ZLIGSYE?<].Y@^CAYU,  
M:B<CH@\_#R8@0T7`R9AX^#8AHRW`R9AX^+\$6X95@9\$4893L8LE9>DF3L9\$;H9  
M3D8<AI,Q`^9.1L1DAI,189CA9\$3MPLFXR<EXP<GXX"O)/.C7\$7`N=GB\_>6`W  
MJ<S7!L?]&%7T`I^2JG\_1?R?3&+Q23F89T!9`QVZ%H9WG3PZPCKEX@!4])#:`  
MZLE)3+VBX==A.,K?Z<ZE\,HYEZ4^[EQ\5ZMSZ7WP=?6`/P\$0^1J<]\\*T^J#J  
MJ\$Q0`N8:IS>%?63F[ ]Y\*6U[0U-,: ]X>;^097545#\_NK87\_5D[^:~!>?\_BY-  
M9]"[0^\_HW\A@:A&OG@?`HUWVX^/D=U3&P]D\^:IZ/T6CN-GN)N-;^?LI?!]E  
M2?`!+02,KI]XQ/HXSIN9TW^M^AN9]KYHR-..][<>G00.]Y4X?F?5\;&9KW75,V  
M-.\[4CXT[W0BM\*%YW^,JAN9]ARJ`YGUG5`W-^W97#`W[\_GR(W0>%FJ\_V?7\%  
M\_V,)N+<03WQX05SV(352[!\_C9Y=^]MKV\*` (T45^GZ^>:0^BAZ76+\_8N\IP>  
ME^~9H/.@A^`^]-AV`Z;K] ("N%67M>QPQ)S/1Y8&]W(M83,%R\$`XOTF^QIQ`K  
MQ\RB:\*VHE>^(6P2\U0V=JKY;+D=?%?S`JHNEJF:0NDW7TU`\*C#B[->3)TVD7(  
M]0G:2AT[\* [&\$5NN66+?SCDCLB,3V2VR\_Q'9);)?\$-DILH\0626R1Q"Z:VL[[  
M(\*1^+K\$+^A5BFR2V26\*W2>PVB>V5VZ)/2&Q)R3VO,2>EUBWQ+HE]J;~\$WI38  
M]Q+[@0&\*J(C:5WD>NQ` (E-D5B4R1VG<2ND]@B:V06\$QB,:40-I5AL8DR?72:  
M]P/CZ#~LRLN\?;+90NC?G+@S@>5\*Z^=S@J)/C%;K'&&Q7G04]CG#.A,H-!-,  
MHIG@`~8\*1M9-[#[/#78/C,4]DH;7VP!K!%H+~P&0(\*&U[090Y!M?P0X`@7J  
M8Y2IG>3RT04#4/:Q\_4#9QP[ `K\>A/`\$EEWC\`T(Y`'\_`F;/`.]AX/IK=I`=  
M`1\_5!>CS@`:/-XVN?`\*>97\YV-`>QDDNX%7T/`Z7T#^#:CM7T`GO[,`86P;  
M;<7KLU#>`=I\_0WT?0`V"AM>?P"[CBU#0%^R`T#='Y`W@]?#^`^`]]W7#^`^\_  
MB/OON=1I1ZG325\*Y4BH7QF(JJ68E-\4(NU.7RH.K9`G:ZB(IAM1)\*7724>ID  
MDE2!(J0\*4K202PF\*(1674G%`J;@B+(^6KQ9\_9/=9GM,9<=YT+]"V`\_CQZ%P  
M.:?.C1C7@%=VB^:~1[%8J^`SV^0\$`9.IG]82Q;!6PR8G2)MTDJ I+DJJ7406.  
M405)4DU2JLE1JBE)JD-\*=3A\*=21)=4JI3D>ISB2I+BG5Y2C5E235+:6Z':6Z  
MDZ1.2ZG3CE^GDZ3RI`6=26E=W?1QC-VZ\J1UG4EI755\*;EUYTKK.I+0N+F6W  
MKCQI76=26A>7\$134]N0V68-A(5;\_&(1?GZD?ZD`N,>G%:~LT6QKCFPCVSM>  
MN^&[JJ`"SXZPE?%N^&B`-95&NW`X;`\*OPSAFM)T38C9([?4-FI\$?Z)OPDS5  
MGT-8^E3K,]<8^EBE5%8HI0I32.6`G`HI0BI\$>]-0\*N105P\$+,9=-R@4M\$`.^  
MYE"72U)\$WYA[V3J/&?T\0&?TK<DHZ`]J\$=AY[ ]<ITR2E`"AUK\$I2YD0`/\*#4  
MPX=@@K)640X4\*\$VP=A.4`9+2#)0.%I64>R7E`J]TLNF2<E!2`@=%ZN5E!<D  
MY7F@=+,9D0\*JI/P2\*\*93\$EQP>>>2)D(Y4X847/ARP]!>9HHF&0Z`90Z2&`)  
MRL=\$`364`0"JUYLH=0H]7W@.<:~#;`PH?^R`]JQ,^]A55+5IY>VM;K)Y>XJ9  
M/\$FVE>RKK+;U3[':UJ;KN2;?96Q0C`Z`S5[,~AC61T`QW80<\*0%I,"~@`?CG  
MI^\*G;V60`RBMK6NC%80:N\$)K\1FU41T78VL1S325%145?IV2HX^M8K)6L;DU  
MP#!ZQ!HS@<>KZYW/7J3ZHA\$<E1Y9WWSF43A>;<-7\*U9^T5>Y^K@MIA2?L;75  
M3\_7BW;IH1Y\_1PK40+V(+\*Z='\*MJHE5^7:/U/M9`G\*955=7.F.`(5:+TI&N4  
M;C0%])62:>,EX^USZ\_R\9.M:=^@MK(A\$JT^KU\*K33]\$8<KU%XZLB,KTR6CDC  
M9?MVZ.USTF0=GZDS=P&6:L66;LH.6J.,FG+=18J\$[/663^CA8Y>5R#\_~X3CS;  
MWF0F>Z<;,Q1]UQTD.\_\$S\$P5Y56[\$;\$^..W^W[I=: ,ZV#D9[W"\*RQ#^0>6I?P3



The data there will be the real and imaginary parts of the differential phase decoder output for every carrier. The plot will have the periodicity of 30 because there are 15 carriers and 2 values/carrier. You can see there how much the carrier amplitudes vary and how clean the output is: for no-noise output the imaginary part (the second number) should be zero. I hope to find out this way whether the problem concentrates around the edge carriers or maybe it is spread all over the band.

Pawel

---

## 50. FW: A piece of my discussion with Joni

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 11:18:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30418C69@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

-----  
From: Pawel Jalocho  
To: Danie Brynard; Frode Weierud; Timo Sivula; Johan Forrer  
Subject: A piece of my discussion with Joni  
Date: 24. August 1995 23:09

Just a piece of my discussion with Joni, I thought it could be interesting.

> > Does your NOS behave the same ?  
> >  
> Sorry to say but yes. We must do first some coding to get this one work.

Maybe at least Timo's T4 can issue KISS commands ?  
I found a partial way around this problem in JONS - you type:

comm tnc "\300\10\1\300" (this command sends a given string to a given interface)

In this example I send the following characters:

C0hex, 8, 1, C0hex and the comm adds CR (13) by itself.

C0hex is the "new packet" flag in KISS protocol so after all I send a complete packet saying: set parameter 8 to value 1.

The problems are:

1. I cannot send a zero because a zero byte is the end-of-string marker. Thus I cannot set the FEC nor the Interleave to zero !
2. The "comm" adds the CR by itself thus infact I start a new packet and put one byte into it... this makes problem sometimes and the first good packet which follows is not accepted.

> Yep, but it works well with FEC 3 and we tested this spy thing you  
> tols us to do, Timo had problems with his spy and it made 8 MB files on  
> hid HD ;-)... user error who knows ;-)

This is very funny... for me the SPY makes a file of 512 lines and there is one decimal number in every line. I never had problems with that.

> > You could try to set transceivers to some low power (5-10W) with FEC=3  
> > and then ask somebody to put a dead carrier on the channel  
> > - the packets should still be received !  
> >  
> Yes this test we MUST do ... very very intresting one.

The result maybe mixed because already some carriers seems to be very weak but you should try.

Then, if you set large Interleave and FEC=3 ask somebody to make burst noise or make it yourself with a bad lamp switch :-)  
The packets should still come through...

With FEC=3 and Interleave=16 you can withstand  $3 \cdot 16 / 2 = 24$  symbol burst error - this makes  $24 / 83.33 =$  about 0.3 second.  
A good example of a burst errors is when you hear a distant lighting storm  
- the static crashes can blank any signal for a fraction of second  
- well, sometimes for a few seconds :-)

> My results: with normal usb and width/shift knobs as they were during  
> fec 3 testings i got results between 2.1 khz and 2.3 khz !!!  
> I know that with width/shift knobs i can change this a little.

For me the points where the S-meter falls down by 1S are:  
29.999.6 and 29.997.2 thus I get 2.4 kHz at -6dB.  
The passband center is at 29.998.4 which corresponds to the audio  
frequency of 1600 Hz (if the transceiver is aligned right).

I have placed the modem's center carrier on 1625 Hz - so for me  
it all right, for you... I don't know :-)

The modem in principle needs at least 15 carriers \* 125 Hz/carrier = 1875  
Hz.

Because the symbol shape is rather short it spreads a bit more over  
the frequency so it would be better to have about 2100 Hz.

The problem is that you don't know where this 2.1 kHz is located in your  
radio (where is its center). In the receiver you can adjust the band's  
center but for the transmitter you probably cannot.

On the diagram of the FT757 I can see that during transmit, the IF shift  
is forced to a certain voltage regardless of the IF shift knob setting.  
Would be nice to have a tool to measure the transmitter response and place  
the signal's spectrum at the best position.

I could let the DSP send a series of tone of different frequency  
and you could watch the power-meter for amplitude changes. Then  
you could find out the passband edges and the center ?  
This sounds like a good test to let us understand where is the problem.

By the way, you can try the LSB mode - I am sure the response is going  
to be different :-) and the result may come out better.

Pawel

---

## 51. FW: Various comments on the newqpsk

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 11:17:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30418B9E@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

-----

From: Pawel Jalocho  
To: Joni Becklund; Timo Sivula; Danie Brynard; Frode Weierud; Johan Forrer;  
Jarkko Vuori  
Subject: Various comments on the newqpsk  
Date: 23. August 1995 21:57

Hi to all !

Joni just told he made some successful tests with Timo !  
He asked some questions and I thought the answer would be  
interesting to all people involved...  
I hope Joni doesn't mind that I have included his words here...

>Modem works best when fec mode 3 is active. We tested it without  
>fec and it worked with about 30-40% repeats so we think that ft757  
>filters are too narrow for it anyway...

FEC=3 has best correcting capabilities, so this is no surprise.

The reason for many errors with no FEC I can see two:

1. the filters are too narrow so the carriers on the edge get  
many symbols corrupted.
2. The spikes of the modem's waveform are cut off by the transmitter  
and this makes lot of errors on all carriers.  
The only cure for now is to lower the driving level - this lowers  
the output power too...

>Interesting point is that fec mode 1 didn't work at all.. dcd flickers etc...

This I don't understand at all, please see if the modem works with FEC=1  
in loopback. I could have screwed up something with FEC=1 in the recent  
version. In principle if FEC=0 works then FEC=1 should work too  
and much better.

>With fec 3 we had nice qso and we down/uploaded some files without  
>any repeats. Signal levels during test were 59+ - 57 ....

You didn't do very high speed but it's a great thing !  
In next tests you could try to lower the power and see how much  
you can go down before you start missing the packets.

>We tested modem also while txjam was 4 and txtunelen/txsynclen 16..  
>  
>It worked fine like this also.

I understand you tried to lower these numbers for lower overhead  
due to preambles and post-ambls.  
Did you modify the corresponding receiver parameters like RxMinTune ?

>Timo has some problems with his receive, he said that dcd flickers  
>when packets are received... Mine worked like this: at the beginning  
>the dcd is steady the it goes off and comes on again and says on  
>until hole packet/s are gone. So i think this dcd-off state is when  
>txtune/sync off and packet starts ?? Im not sure but with short packets  
>this point is abt half on hole packet lenght.

I should explain here the details... when the receiver detects  
the tune preamble it turns the LED on. Then, it keeps the LED on  
while it receives the symbol-sync preamble. Now, when it goes into  
data more the LED is switched on or off depending on the FEC decoder  
errors.  
Note that just after the receiver switches to the data mode there are  
only the four sync. tones active thus the received data is mostly garbage  
thus the FEC finds error and the LED is off. Then, when all 15 tones come up  
there is still some time left before the interleaver fills up. Only after  
some time (propotional to the "Interleave") the FEC decoder gets a  
reasonable  
data so the LED comes up again.

I would say Joni's LED behaved exactly as it should and no flickering  
ment almost perfect data. Timo's red was flickering - thus this was  
either his received (audio level too high or too low) or Joni's transmitter  
drive level or something in the filters or the frequency error to high ?  
IF shift maybe on Timo's side ?

Timo could try to use the AGC - I think is pretty safe if you don't want  
to play with setting the gain right. Joni could try to lower the transmitter  
level.

>So I would say that is WORKS !!! but I must say that it would be instesting  
>to measure ft757 and its receiver on sbb... I mean how big differencies  
>there are in signal levels when testing hole sbb bandwidth.

>From what I see on my FT757 the ssb filter passes the audio from 200-300 Hz  
to 2700-2800 Hz at -6dB. I measure this by turning the calibrator on and  
tuning between say 29100 and 29103 kHz and observing the S-meter.  
This is not bad at all... but, the audio filters after the SSB demodulator  
make a strong dumping of high frequencies, the band 500-1000 Hz is passed  
well but the 2700-2800 Hz band is attenuated by about 3 times (9dB).

>ps. maybe we could to some testing to africa later ;-)

We will all hold our breaths :-)

Pawel

---

## 52. FW: Various comments on the newqpsk

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 11:18:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30418C1E@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

-----  
From: Pawel Jalocho  
To: Timo Sivula <Timo.Sivula@ntc.nokia.com> Danie Brynard; Frode Weierud;  
Pawel Jalocho; Jarkko Vuori; Joni; Johan Forrer  
Subject: Re: Various comments on the newqpsk  
Date: 24. August 1995 10:00

Hello,

> We tried reducing the mic gain at Jonis end without any change. We used  
> txattenuate 20 dB. I noticed that the best recption I got when Jonis TX  
was  
> on  
> very low power, he had his microphone disconnected and I had the RF gain  
> attenuating  
> so that there was very little noise coming trough and the level low. What  
is



> the level coming  
> out of the modem? Could it be possible that the signal gets distorted at  
the  
> TX  
> already \_before\_ the mic gain potentiometer?

I didn't measure the modem's output level but it should be around 500 mV  
and the 20.0 dB attenuation would divide this by 10 thus making 50 mV.  
If you think it is too much just put more attenuation... 40 dB = 100 times  
=> 5 mV output. The thing is I don't know how much noise is present  
on the CODEC's output...

> I tested both with the AGC on, and without it. No big difference. It is  
> clear that there  
> were errors in the signal coming through as the red led flickered when  
Jonis  
> modem  
> transmitted. The QSO was perfect however, because the fec corrects these  
> errors.  
> In real HF conditions I do not think that this modem would work very well.  
> I agree with Joni in that one probable problem is the receiver bandwidth.  
I  
> tried  
> playing around with the shift and width knobs but they did not help the  
> situation at all.

The width knob ? You mean maybe the notch knob inside the shift knob ?  
This one should be switched off - otherwise it notches out a piece  
of the passband.

To judge the modem performance you could make a comparison test:  
With FEC=3 see at which power you can reliably communicate.  
Then try to make a packet QSO with a standard 1200 bps modem  
(maybe you need to switch to FM...) and see again how much  
power you need now ?

> The only thing that made any difference was using the RF gain to attenuate  
> the incoming  
> signal, this improved the LED behavior. This would imply some form of AF  
> level related  
> problem.

Wasn't the noise blanker turned on maybe ? It could clip and blank pieces  
of the signal. I would try both slow and fast AGC setting too.

> FEC1 seems not to be working very well, I have not made any loopback tests  
> yet. I think  
> that the error correcting is not powerful enough to correct the errors  
that  
> FEC3 manages  
> to correct. I could copy a few of Jonis packets but he did not copy  
> anything. There might  
> also be something wrong with the algorithm, too. Have you made loopback  
> tests with it, Pawel?

Yes, I have done tests yesterday and FEC=1 makes a much better copy  
than FEC=0 in the presence of noise. I agree that FEC1 isn't as good  
as FEC3 but what I don't understand is why FEC1 is worse than FEC0 ?

> The modem worked great with FEC3, the qso was like any packet QSO, not  
> better not  
> worse. I would say that the improvement to the earlier versions is huge.  
> Congrats Pawel,  
> you are coming very close to it now! It might need be wise to maybe remove  
> one carrier?  
> This would squeeze the thing a little and it might fit better into the  
used  
> bandwidth?

If I remove one carrier, we can only work with FEC=0...  
I see if I could easily remove maybe two "edge" carriers.

The other reason for our problems can be that my front FIR filter is  
too narrow... this would cause problems when the transceivers are mis-tuned  
by too much. Please try the next time to change slightly the frequency  
and see if this improves the error rate.

If you connect two LEDs to the UP/DOWN outputs (next to the PTT output)  
you will notice if the modem is mis-tuned by more than 20 Hz and in

which direction. Better yet, connect these lines to the up/down pins of the microphone socket of the FT757 - and the DSPCARD4 will should correct the error automatically at the end of every transmission. This have never been tried however :-)

```
> >>ps. maybe we could to some testing to africa later ;-)
>
> I've hear rumours of this. Could somebody inform me a little, too? I would
> very much like to join these tests.
```

Just make an appointment with Danie !

A question to Timo: with T4 can you give an arbitrary KISS command ?  
I am working on switching the FEC and Interleave via a KISS command so you can do it on-line during a connect as to quickly find the best setting. For eexample I would made FEC switchable with parameter 20 and the Intnerleave with parameter 21. Other things (TxAttenuate for example) could be made switchable too.

Pawel

---

### 53. FW: Program to test transmitter passband

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 11:18:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30418CAD@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

-----  
From: Pawel Jalocho  
To: Danie Brynard; Joni Becklund; Timo Sivula; Johan Forrer; Frode Weierud  
Subject: Program to test transmitter passband  
Date: 25. August 1995 0:48

[[ TXTEST.ZIP : 2398 in TXTEST.ZIP ]]  
So, I have written a simple program to "measure" the passband of your tranceiver's transmitter. How to use it:

- Compile the source and load the .LOD into the DSPCARD4/EVM56K.  
You will hear a tone in your headphones.
- Start any terminal emulator, set it to 19200 bps, 8-bits, no parity and the serial port which is connected to the DSP's SCI.  
With <,>,[,],{,} you can change the tone's frequency and you can see it's current frequency on the monitor.
- Connect the DSPCARD4/EVM56K to the transceiver (PTT and MIC lines).  
Push "p" - and the DSPCARD4 will key the transmitter (dummy load !).  
Set a low power level (5-10W) and the meter into such mode so you can observe the RF output power.  
Pushing "p" once more switches the transmitter off.
- Now you can vary the tone around and see how the output power changes.  
Notice the edges where the power quickly drops out and notice too how the passband looks like, whether it is flat o sloppy.

My passband begins at about 400 Hz and looks flat up to 1000 Hz, then it falls down slowly but systematically. The upper edge appears at about 2700-2800 Hz.  
I must say I don't like this... for the receiver's non-flat passband I didn't care so much, but if the transmitter makes the same, the upper carriers are going to be much weaker on the air than the lower ones and this will truly affect the error rates. What can we do about it ?  
For now I can see only one solution: pre-amplify some carriers before transmission (a very easy thing to do).

I suggest that now everybody should measure his transmitter and put down the output levels for various frequencies  
- ideally for every carrier being used by the modem.  
If this is done we can put the numbers into the newqpsk.bas as to generate a set of tones which would have equal amplitudes at the antenna socket.

Pawel

section 1 of uuencode 5.02 of file txtest.zip by R.E.M.

-----  
--  
This uuencoded part of the message containing the file txtest.zip has been decoded and converted into an attachment.  
-----

```
--
sum -r/size 10413/3479 section (from "begin" to "end")
sum -r/size 38585/2505 entire input file
```

The following binary file has been uuencoded to ensure successful transmission. Use UUDECODE to extract.

```
begin 600 TXTTEST.ZIP
M4$!#!0``@(`,,$&1]>H^">4PD``)@@````*````5%A415-4+D%33>5:_U?;
M.!+_^7B/_V$>[3V@F&`[0$N@W%) (KKD-)00L+00N[0%D6TE$;<F59)+LW?WO
M)\EV+.<+I3WVI_-[-+8T\)%H9B3-C'K<Z?;+:B]NVW?W#;.:BX=Z"$A`4=$
M$D9;X._+$;`!G&7#3+5[1T<'ZVO'@B1IC$%R1$5"I,0<)%;=DD&$U5="J.H=
M510X`I1%A$&*A"@0C13(5KAM*/QCB&OZ&8A2/DP$W_Z)?K<T=1X-T$D;@%
M#WG7#`&(180VR."A@:.LD<8.B/3HD8<_#,<-]?8YF#90D0(&XT/%0KZF_GGS
M4H""EF2DAC02I)@"!D5EI]##M7RX/#G\ $P%\R<*%ZCEUX?PH7-_WSL^N+
M_?4U6/$<*F5&&*!0,JZY/V2#`J;5P)K;.[0$;.^Z:)]0"@BRP4"959#?L>:[
MELDTDX`**F5&&*!0,JZY/V2#`J;5P)K;.[0$;.^Z:)]0"@BRP4"959#?L>:[
M)L.1/!\A2I6I"C[WI15<Z4>$XB%)08P8EW8[93$1JN5/A(9QF`8C#&C)-I4
M37F/DNB&*?,D*.1,5+QD`/8<UM?ZM[>FPQ!`PB)<:;$!WQ(3B%HOQWQ-8[H6C
M825""\`]?#^VK?N6N[G3:EE4QC9+R"\<`_`^N-4@Q_6>9ME3&DYI$`[:'W[(`
M*QA=+I6)1F3PXJOB=H0!A3)#<0RAEFV+4(DY1?>$GN00D'(VY"B!Z[/+;=N>
MO:0+!B7T6Z]=53-'[M%L]?I=&QZQH>0MC*!^;T>JMY%X-6?-8[KN:[#$EXH
M!E,4Q/EV)/2^>'UU"5(WV=P)>U1*_6"<?L=W^-LE73U,W^SO>D[R=A6GP_WY
M#LU`_:?0?!LME#P.(_`4DI,3.;WNIW;7<1NN^=-?5`_V#Z[Z#065F"65I)
M,55HU5ZRYEY]Q/G8-VH]ACB@:.`Z-Z2]-T$D,0L_#S^EJ?LQ`+T6_LG9`_
MJ>*>@JNU!<#CY54'E(WW(V>=X[EHHYXCW1TA@9Y)O5\=#+(UA.`XQ><0</IU?
M;0H($>=$?:6;>"F,GHR#2M)'T2.B86[C.1X413!Q;6$="-O(<2YZYF71_6MR@
MM6XR19HN!K13["5L(@,":!`G,"9.%,`NT[B.M3=7ICY0>W=^8%<`YK?4M[
MT;:^=HU1U*5*EHH@9NQ56#MW2KW0*80CQ$H,1? *QODJP)SH-<FXA+]84P\%
M%Q@FGZ$Y9$F]JT`Z)(U<[J9;MKJ#16Q5KCN.=V<V0B.01%"RA4ZCFKCXB]J
MW-0;V@9HHYS444Y7H]Q(G`H_I\MA_EF'.?D*S`4;T^5`?Z_-=;AKQKX/ROH
M5X_PK^4<$64C+*/+D:T>17M26\WF:$(!)]`G44WUAMCRL06U0JV+ZV_ITC34
MSL3]^AIT*JP%60F5>?L28WRO'"(+OE^,7!+_F]3@OZP:_.]5@_2:FA^DQJ:
M+ZN&Y0>JH?F2:K`^5DEBCY90EM9)[#=<<QKKXWCB683IE&NY)MZ":*_V>K.
MG55&C$]9LK@MPV9-@#23X2+1Q^<0_X,(J_Y!,WBE0)R$>@L=L$@?X?"5%8!4
MIR"\AQ$;0Z*/0V$(! (PQC!&U#E$D8D#VX<E]Q3<>88YSXB)0TB>HB8]TMBLD
MX@L8.CBR<6H<WE0B:)IH7NI[;XZ,OUW@Y%];SP]I#"<6'GV8+$9P90RZ='6
M!V61"(XY2A%G&8T<X'GLPCB#`8@Q'8XDG"JM<"Q&+(YJ,ZJ/:TXL5\8>:R"%
M5N,ONJD4]6B&4<ASVY;6G5XA@)1QBD`L>B*[[@Y,2#E];)]Q94W-)S*N(TD
M6T_EY0UM<)YS(*DFK,LC9T#Q&(AVYIE#J<X[$[D]@:;^)(-`Q[OH*7G]%;/&
M+#0)01DMCQ,5X-,#JM'&7-=H3$2POC87IE>T,P]'A7^SP6QR`\X2X_FF//#$
M@`F*`<$7=:U]]ZEYL"MC0X$'\>2//A)\`4$8V>E4V?M:R73;ETH&J]5PM6K-B
MQ=PF.K\([(:`XM#.`>XZEK<+[N)8;&17:PIB&C$H2`](3DYCS+)4ZT\)?$!+>
M"4QLOC#L0%Q_P,KY]?]ZK?2[595,/V&M^='/'F!\V#=#T=OCYI^]W^H3G/
MY@(XK+I$BD,ZDGJEYUT6'/5!Y21#ADZG58JT8H[C&1(T4L<,AH!(Q'F,6
MK@A00,E0Q%'4EE@&U":<C8A25[6]"`<8TSS:@:DC)AJY)+LM3[1B000`=&A
MXMF"74]][:[#NSHMYU^=WM6:_J^HLK<,>?/ZG?SWQ2-&9""U\([U:F,),B0
MD@)940'M5)NFTC(%`8W0ZB\`41.?:ULE%K9-+L6AY;1[&K8/MU<GK?;<6
M3)55">X6,]3^JJ>W0E:]F!DH:F,Z-A@(+)^7^)+#SH*XS+4$%>L1P2-!A&I/
M*;!@:D=V8<SAE=]4C,X-H;=:L?<Q?01M]RW*`JXN"ZP@,B3:]T=ZFZ7592
M!4*&N,%"ARYCVLX--V/R*J8Y%A.S>EI<;:]^=_&RH>5M;F!R4!D#U>!>%
M'`8ULQ4:E&';(%)-RT;36G4"J"2UW!;12/NMIQ_7\V9^>QP1\56_-8)Z&CI8
M%#27,S"3?*/GV7#W#@.FOELU0A!*?42F:=NH2&TZ,W(L1U@IEX+VT53A=5
M2!8J#$JS&16*+,5A2(MU0>VZH,Z7YUK_G2I_,.9/0-8J6+LHF;VK7P)FI`D
M2P!SSC@0`2C0^XK;>-O>/7P&/Y[@, #-[J+G+V=VW-_9A]!$".$.TC.W#>25(
M?@>AU`!&438TRV.#;$4;2DT;LSSQDY)U%4RJ"0`!S9)`9Q0]1[AD`0HUN%8
M\@Q)ZH6]_`BOV!HZ6@[LRLR%;M!^Q9R@B=I^ANB""GD:8HK];2TM+/Z[E+
ME62ZF\X"=CVK?]*F&D`J]U00(RLUSA01YLPW<P#2*:P[^AZ2HQ]WB7@0B)&^
M"UF\`"KG+KT*<WXK?W0*K$4/6Z>1DZK>7_Y?E=SE5R5Z.),AYW4)@"B$"+@
MC=Z_JJ*$(3$5<4WBFD]31C,<R\E-?2E`7$W0+(9<0K&^-EL<D0#OQ: ^GVI,G
ME$X&^:7K[[:75"SAEY;P<DOX-4M8MFAV<EN8WU[^:UW!Q?8-@VUIKX[0>BOP
M03E\KXY?7NNMG)/VA3GYZT)5!`MWA?G-EC9.`F7)X(^(^,7ZOP:LK_T74$L!
M`A0`%` `` `` `@`PP09`UZCX)Y3"0`F"````H``````````0`@` ````````%18
>5$535"Y!4TU02P4&` ````$` ``0`X` ````>PD` ````
`
end
```

## 54. FW: Various comments on the newqpsk

From: Sivula Timo <Sivula@ncsmg01es.ntc.nokia.com>  
Date: Mon, 28 Aug 95 11:18:00 eet  
To: "'HFSIG'" <hfsig@tapr.org>  
Message-ID: <30418CED@ntcite01es.ntc.nokia.com>  
Rendered body type: text/plain

-----  
From: Sivula Timo  
To: Pawel Jalocha; danie.brynard; Frode Weierud; Jarkko Vuori; Joni; Johan Forrer

Subject: Re: Various comments on the newqpsk  
Date: 25. August 1995 9:35

Hello!

First: Should we either move this discussion to the hfsig group, or create a new group for DSP56k qpsk testing and reporting? I suggest this, as I am getting fed up with this carbon copying every time :-)

I will comment the latest three messages from Pawel in this one and only message, as I think it is easier and the topic is of interest to all involved here. The message is quite long, I am afraid. Maybe I should have kept it in three parts anyway... I have some really weird ideas presented in the end of the message...

-----  
>Subject: Re: Various comments on the newqpsk  
>Date: 24. August 1995 10:00

>If you think it is too much just put more attenuation... 40 dB = 100 times  
>=> 5 mV output. The thing is I don't know how much noise is present  
>on the CODEC's output...

How does the headphone output relate to the line output? At least the headphone output is quite noisy. Do you think that 50 mV could get distorted in the mic preamp?

>The width knob ? You mean maybe the notch knob inside the shift knob ?

Both Joni and I are using the FT757 GX MK I and it does not have a notch, instead there is a shift function where the MK II has a notch.

>To judge the modem performance you could make a comparison test:  
>With FEC=3 see at which power you can reliably communicate.  
>Then try to make a packet QSO with a standard 1200 bps modem  
>(maybe you need to switch to FM...) and see again how much  
>power you need now ?

This seems like a good idea, the only problem is that as far as I remember Joni reduced the output power until it could not get less, and it still worked ok. The 757 has quite good dynamics in what comes to setting the TX output. Remember that we are only 10 km apart.

>Wasn't the noise blanker turned on maybe ? It could clip and blank pieces  
>of the signal. I would try both slow and fast AGC setting too.

No noiseblanker was used, and I tested both slow and fast AGC.

>Yes, I have done tests yesterday and FEC=1 makes a much better copy  
>than FEC=0 in the presence of noise. I agree that FEC1 isn't as good  
>as FEC3 but what I don't understand is why FEC1 is worse than FEC0 ?

This is what I suspected. We did not do any long tests with FEC=1, so our report is not 100% valid. I guess that it simply did not manage all errors that were on the signal. FEC=3 could. Or we didn't tune it long enough.

>If I remove one carrier, we can only work with FEC=0...  
>I see if I could easily remove maybe two "edge" carriers.

Too bad.

>The other reason for our problems can be that my front FIR filter is  
>too narrow... this would cause problems when the transceivers are miss-tuned  
>by too much. Please try the next time to change slightly the frequency  
>and see if this improves the error rate.

We did tune a lot, but there was no big difference in the result.

>which direction. Better yet, connect these lines to the up/down pins  
>of the microphone socket of the FT757 - and the DSPCARD4 will

>should correct the error automatically at the end of every transmission.  
>This have never been tried however :-)

I think this is the next step we will test. I'll have to find out a neat way to connect the up & down buttons to the rear of the rig, so that I will not have to connect it to the mic. An other solution would be using a mic plug to feed the AF in to the rig, not the AFSK/PATCH socket in the back of the rig. This way I could connect the DSP4 directly to the mic socket. I even have a spare microphone plug that I could solder in. Hmm, I think I'll try this. Does anybody know if there is any AF out in the mic socket of the 757, so that I could connect all lines to the one and same plug?

>A question to Timo: with T4 can you give an arbitrary KISS command ?

Sorry, no. T4 runs on a TheFirmware TNC, and I have to use a KISS emulator (TFKISS) to interface T4 to an non TF TNC.

>I am working on switching the FEC and Interleave via a KISS command  
>so you can do it on-line during a connect as to quickly find the best  
>setting. For example I would made FEC switchable with parameter 20  
>and the Interleave with parameter 21. Other things (TxAttenuate for  
>example) could be made switchable too.

Why not make this automatic instead? Let the modem collect the number of errorcorrections and then according to some trig-levels switch automatically to a less complicated FEC and higher speed if the link permits it, and vice versa.

You only need to find out how to tell the other modem about the mode change... :-)

but with 15 tones to choose from that should not be too difficult, I hope.

Now the modem would autotune and fit the errorcorrecting code according to the band conditions, what say?

Now to the second mail:

>Just a piece of my discussion with Joni, I thought it could be interesting.

>> Yep, but it works well with FEC 3 and we tested this spy thing you  
>> told us to do, Timo had problems with his spy and it made 8 MB files on  
>> hid HD ;-)... user error who knows ;-)  
>

>This is very funny... for me the SPY makes a file of 512 lines and there is  
>one decimal number in every line. I never had problems with that.

Hey guys, tell me how to use this spy-thing! Here are the original notes by Pawel and my comments:

>Timo will do the following:  
>- compile the source with SPY=1 and SPY\_RxCrossedIQ=1  
>- disconnect the PTT and MIC lines from the transceiver  
> because with SPY=1 the modem transmits random data packets.  
>- load the .LOD file into the his DSPCARD4

All the previous went fine.

>- start the SPY.EXE (supplied with the DSPCARD4 tools) like this:  
> spy -p2 (-p2 means the DSPCARD4 is on the COM2)

I have my DSP on COM 1 so I used SPY -p1, and I tested also just SPY. The program started ok.

>- press "s" when Joni's beacon comes and the red LED flickers.

I pressed s once and there was indeed a funny colored figure on the screen.

> After a moment Timo will get some funny data displayed on his screen.  
>- press "w" to save this data into a log.dat file.

Here came the Problems. I pressed w once, and the hard disk started running. And it did not stop, not until I forced the system down with ctrl-alt del.

When I then  
checked what I had made there was a 9 MB file calle log.dat. This I can not  
send  
to you ;-) What does the S do? What happens when I press w, or what should  
happen?  
How does this SPY work?

>By the way, you can try the LSB mode - I am sure the response is going  
>to be different :-) and the result may come out better.

This we have not tested with the latest version.

>So, I have written a simple program to "measure" the passband of your  
>tranceiver's transmitter. How to use it:

I will be out of town for the weekend but this test seems to be a good one.  
I'll  
run it when I get back on monday. Could you make the test soft so that one  
could give the modulating tone directly as a number? Like 100 (enter) would  
give 100 Hz, 3000 (enter) would give 3 kHz? This would be very useful for  
other applications as well... A settable sinewave generator on the DSP4!

>My passband begins at about 400 Hz and looks flat up to 1000 Hz,  
...  
>I must say I don't like this... for the receiver's non-flat passband  
>I didn't care so much, but if the transmitter makes the same, the upper  
>carriers are going to be much weaker on the air than the lower ones  
>and this will truly affect the error rates. What can we do about it ?

Pawel, excuse me for my mad ideas but... Could this be done?

Make one of the preamble tones being in the high end of the pass band. Like  
1500  
and, or why not using 3 or more tones.

The receiving modem, during tune phase, measures the level difference  
between this  
tone and a tone in the "flat" response area. The dummy bits (pre/post data  
or whatever  
they are called) is then used to send the measurement data in the next  
transmission  
back to the other end.

The modems would monitor the upper end of the AF spectrum all the time and  
adjust the  
levels according to what it gets form the other end. I guess that  
quantizizing with 4 bits  
might be enough? One could assume that the passband is flat to 1 kHz, and  
quantize the  
difference between this "flat" level and the high end level. If one asumes a  
linear decay  
of the response from 1 kHz upwards only the highest tone has to be  
monitored, and a  
linear correction from 1 kHz to the upper frequency would be calculated. If  
the linear  
assumption is too inaccurate, why not measure more tones?

>For now I can see only one solution: pre-amplify some carriers before  
>transmition (a very easy thing to do).

How about automating this, too like above? Is it possible? How much  
processing power  
would this correcting need if it would be done "live"? This feature could  
also help against  
selective fading in some cases on HF. The correction could of course either  
be done at the  
receiver or the transmitter end, or why not both? I would prefer the TX side  
as this would  
increase the S/N of the higher tones.

>I suggest that now everybody should measure his transmitter  
>and put down the output levels for various frequencies

Yep, and if a common knee point (1 kHz?) is found it could be used  
as the other end of the linear correction... :-)

What do you think?  
73, Timo

## 55. FFT progress

From: Phil Karn <karn@qualcomm.com>  
Date: Mon, 28 Aug 1995 15:55:55 -0700  
To: hfsig@tapr.org  
Cc: antonio@qualcomm.com  
Message-ID: <199508282255.PAA22381@servo.qualcomm.com>  
Rendered body type: text/plain

I thought the group might be interested in my progress on FFT code for the Intel machines. Franklin Antonio N6NKF and I are in a little friendly competition to see who can do the fastest FFT on a 90 Mhz Pentium. He's using Windows and the Watcom C compiler while I'm using BSDI 2.0 and GCC 2.6.3.

So far Franklin has been a little ahead of me. From looking at the asm code his compiler produces, it looks like it knows how to optimize for the Pentium, unlike my version of GCC.

So I've been writing some of the butterflies by hand in assembler with careful attention given to keeping the floating point pipeline full (which is harder than it sounds given the baroque Intel architecture).

I now have a single precision floating point radix 4 butterfly w/o twiddles executing in about 0.71 microseconds. That's about 56 megaflops on average counting only "real" floating point instructions (fld, fst, fadd, fsub). By comparison, "straight" asm code written without regard to pipelining takes about 1.3 microseconds.

When I integrated this into radix-4 DIF C code I'd already written so that the asm routine is executed instead of C code in the last stage, the overall time for a 256-point complex FFT is 485 microseconds. (The pure C code was 525 us). That's better than a million samples/sec.

The next step is to write a general radix-4 butterfly that can handle twiddle factors and arbitrary array stepping so it can be used in stages other than the last. This should really get things humping.

Of course, I'm already about 100x faster than required by a spectrum analyzer operating on SSB receiver bandwidths, but it \*is\* fun to see just how fast you can go...

Phil

---

## 56. Multi-tone, DQPSK modem summary

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>  
Date: Tue, 29 Aug 95 15:28:36 MET  
To: hfsig@tapr.org  
Message-ID: <1036.jalocho@home.ifj.edu.pl>  
Rendered body type: text/plain

Here is a short summary to keep all HFSIG'ers aware of what is going on with the multi-tone DQPSK modem.

The modem has evolved to the following configuration:

Modulation type:

There are 15 tones ranging from 750 to 2500 Hz spaced by 125 Hz. Each tone is modulated with DQPSK (four states => 2 bits/symbol) at 83.33 symbols/second. Raw data rate is 2500 bps.

Peak to RMS ratio (crest factor):

Not very good... about 4.0

Frequency error correction:

The receiver can correct a frequency error of up to +/- 60 Hz. There are as well outputs provided for steering the transceiver's UP and DOWN buttons for automatic frequency error correction. This feature has not been tested yet.

Forward error correction:

What is implemented now is basically a block-type FEC with the block size of 15 bits. Every bit of a given block is transmitted on a different carrier. The intention is that a selective fading/interference would affect least amount of bits in a block.

The bits of one block can be all transmitted at one time or they can be spread out in time by an arbitrary factor.  
This feature helps to recover burst errors if at one time all carriers get wiped out by a spark or a lightning discharge.

The FEC schemes are:

Scheme #0: No FEC, effective data rate is 2500 bps.

Scheme #1: 11 "real" data bits + 4 CRC bits form a 15-bit block.  
The receiver can correct one flipped bit per 15-bit block  
Effective data rate is  $2500 \cdot 11/15 = 1833.33$  bps

Scheme #3: 5 data bits are mapped into a 15-bit codeword from a set of 32 codewords with the hamming distance greater than 7 between each two selected from the set. These are in fact Walsh functions.  
The receiver can correct up to three flipped bits per 15-bit block.  
Effective data rate is  $2500 \cdot 5/15 = 833.33$  bps

Hardware platform:

The code is being developed on a DSPCARD4 by ALEFNUL (credits here to Jarkko Vuori and Kaj Vik) but it runs as good on the EVM56002 from Motorola (thanks to Johan Forrer who ported the DSPCARD4's basic system).

Communication protocol:

The modem understands KISS, thus it is oriented towards sending frames. These can be AX.25 frames but in principle there is no real limit here, as KISS can carry any frame.

Some real on-air tests have been done already by Timo and Joni on a short (10-15 km) HF range. AX.25 protocol was used.  
The result... well, the modem works, especially when you use the FEC scheme #3. There are certainly troubles and they concentrate around the transceiver's passbands and driving levels. This you can see better from Timo's postings :-)

A modem version exists already which switches the FEC schemes and Interleave factors according to KISS commands. However, to my big surprise, I found that the packet-radio software I have can not issue an arbitrary KISS command ! For example JNOS:

I say: param tnc 12 3

NOS says: parameter 12 not supported

I have to ask the JNOS writers to fix this bad feature.

Future plans ?

- Double the number of carriers and then I can use the block size of 60 bits which is suitable for RS codes with 4-bit symbols.  
The alternative is to add a block-sync sequence and then you can use FEC with any block size.
- Use longer FFT and better window to make the symbols "narrower" in frequency. The ones I use now have significant sidelobes.

It would be certainly helpfull to have more testers, thus anybody willing to join please stand up ! You need a shortwave transceiver, a PC, and a DSPCARD4 or an EVM56K.

Pawel

---

## 57. Re: [HFSIG:562] Multi-tone, DQPSK modem summary

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 29 Aug 1995 08:45:45 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <BA8FE46747@frl.orst.edu>  
Rendered body type: text/plain

Pawel,

Thank you very much for the summary of the parallel modem. Congratulations is also in order on this fine achievement. What is perhaps not so obvious is



the complexity of this software - for those that are planning on attending the DCC, I hope to devote a bit of time on the modem's software and working principles at the DCC. The system will also be on demo to give you an opportunity to see it in action and ask questions if you like.

Congratulations also goes to Joni and Timo for logging the first successful tests over the air. I realize that developments are in its infancy and there are lots of ideas for improvements.

To all the contributors to this project, your efforts are greatly appreciated - Keep up the good work.

--Johan, KC7WW

P.S. Would you mind if we share your summary with others in the next PSR? I have a few other items that I will include in the HFSIG summary.

---

## 58. Re: [HFSIG:561] FFT progress

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 29 Aug 1995 08:49:08 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <BA9DF76AC4@frl.orst.edu>  
Rendered body type: text/plain

Phil,

Sounds an exciting devolpment. I would not be surprized at all to see some innovative DSP work being done on the faster Intel-based platforms in future. You certainly are making good progress towards that.

--Johan

---

## 59. Re: [HFSIG:551] I measured my RX

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Wed, 30 Aug 1995 06:54:14 +0200  
To: hfsig@tapr.org  
Message-ID: <199508300454.GAA07267@foxbat.pix.za>  
Rendered body type: text/plain

>I made the RX test this morning before leaving to the job. I made it in the  
>same manner as  
>Pawel: I switched the marker generator on and tuned the rig around the  
>marking frequency.  
>If I do not recall wrong, the S meter measures the input \_voltage\_ so that  
>one S-unit is 6 dB.

Thanks for the info. What marker generator are you talking about ? In any case its a good idea. I will use a crystal based QRP rig I have here with good phase noise and measure the receiver passband with. I think my rig can step in 1Hz steps hi !

73 danie

---

## 60. Re: [HFSIG:561] FFT progress

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Wed, 30 Aug 1995 06:54:21 +0200  
To: hfsig@tapr.org  
Message-ID: <199508300454.GAA07277@foxbat.pix.za>  
Rendered body type: text/plain

>Of course, I'm already about 100x faster than required by a spectrum  
>analyzer operating on SSB receiver bandwidths, but it \*is\* fun to see  
>just how fast you can go...  
>  
Hi Phil

Your FFT programming for the pentuim etc is very interesting. I have always been interested in building my own spectrum analyser. What about have a pentuim + radio receiver with say 500kHz baseband bandwidth stepping in 500kHz steps from 100kHz to 2GHz. This would form a magnificent spectrum analyser with unlimited resolution bandwidth ? (sweeps would get slower though)

For having a span wider than 500kHz one could sweep the radio with its RS232 port from the PC. I would suggest an AOR-3000A. It does 100kHz - 2.036GHz at 50Hz increments per second, if I can remember correctly)

I choose 500kHz 'cos it is half your current attainable sampling rate of about 1MHz, not so ?

What do you say ?

73 danie zs6awk

---

## 61. Re: [HFSIG:566] Re: FFT progress

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 29 Aug 1995 23:41:10 -0700  
To: hfsig@tapr.org  
Message-ID: <199508300641.XAA05311@servo.qualcomm.com>  
Rendered body type: text/plain

>been interested in building my own spectrum analyser. What about have a  
>pentium + radio receiver with say 500kHz baseband bandwidth stepping in  
>500kHz steps from 100kHz to 2GHz. This would form a magnificent spectrum  
>analyser with unlimited resolution bandwidth ? (sweeps would get slower though)

Sure! Though I don't know how you'd display 500K points across a typical video screen. I.e., although it could run in real time on a Pentium, the analyzer would display data much faster than your eyes could absorb it.

Lessee...if you drove a typical high resolution PC display at maximum rate, how many samples/sec would that be? Well, a typical display is now 1024 pixels across.

Update that at 72 frames/sec and you have 73728 pixels (frequency samples) per sec. Even after scaling my FFT to 1024 points, it would still take less than 10% of my Pentium 90. That leaves 90+% percent to run the video display driver.

You can make each of those 1024 horizontal pixels cover whatever frequency bin size you like, the FFT computation rate would be the same. You could cover a total of 500 Khz across the display if your RF and A/D converter bandwidth supports that, but then each bin would cover 488 Hz, which might be too coarse for your taste. If you limited yourself to a 10.24 KHz span you could get 10 Hz resolution, not counting window effects.

Phil

---

## 62. Re: [HFSIG:546] Re: Robust modems

From: pe@paccomm.com  
Date: Thu, 31 Aug 95 08:55:20 UTC  
To: hfsig@tapr.org  
Message-ID: <4823@paccomm.com>  
Rendered body type: text/plain

You might also try talking to Bath University - they always had a very strong HF group, unless the academics have moved on.....

---

## 63. Re: [HFSIG:564] Re: FFT progress

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 31 Aug 1995 18:33:09 -0700  
To: hfsig@tapr.org  
Message-ID: <199509010133.SAA26206@servo.qualcomm.com>  
Rendered body type: text/plain

Johan,

Thanks. I've since gotten the 256-point CFFT time down to 380 us, and I think there's \*still\* room for improvement. I'm doing the simple radix-4 butterfly in 24 adds/subtracts, when I know I can do it in only 16 by taking advantage of common subexpressions. ("Simple" butterfly means unity twiddles. I have a separate function that does a general purpose butterfly with non-unity twiddles that takes a little longer.)

The big problem is in keeping the Pentium's 3-stage floating point pipeline full. Floating point add, subtract and multiplies all take 3 clocks to execute from start to finish, but you can stage 3 operations in parallel and retire one instruction per clock if you are careful not to stall the pipeline by using the result of an operation as an operand to a subsequent instruction that starts less than 3 clocks later. This is a fun challenge very much akin to juggling with three balls in the air at once...

Phil

---

#### 64. Re: [HFSIG:562] Multi-tone, DQPSK modem summary

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Fri, 1 Sep 1995 06:43:12 +0200  
To: hfsig@tapr.org  
Message-ID: <199509010443.GAA10424@foxbat.pix.za>  
Rendered body type: text/plain

>Here is a short summary to keep all HFSIG'ers aware of what  
>is going on with the multi-tone DQPSK modem.  
>

....

>It would be certainly helpfull to have more testers, thus anybody  
>willing to join please stand up ! You need a shortwave tranceiver,  
>a PC, and a DSPCARD4 or an EVM56K.  
>  
Hi Pawel

Thanks for this. I am slowly cathing up. I am currently not able to tx from  
NOS using NEWQPSK.ASM

danie

---

#### 65. Re: [HFSIG:567] Re: FFT progress

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Fri, 1 Sep 1995 06:43:28 +0200  
To: hfsig@tapr.org  
Message-ID: <199509010443.GAA10457@foxbat.pix.za>  
Rendered body type: text/plain

Thanks Phil for the Spectrum analyser thoughts. Another solution to a  
personal spectrum analyser for a radio ham would be using a AOR AR-3000A  
(100kc-2GHz) scanner + PC with s/w or the new Trident 1MHz to 1GHz scanner  
which even supports a analog X-Y signal out to an oscilloscope. PC's are  
nasty things in a RF lab. They make a hell of a lot of noise... Anybody  
who's doing that ?

73 danie zs6awk

---

# September 1995

Original source file: hfsig.9509

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-09/TAPR-HF\\_1995-09.txt](https://archive.org/download/TAPR-HF_1995-09/TAPR-HF_1995-09.txt)

## 1. FFT v. MAC

From: Marijan Miletic <s56a@ljutcp.hamradio.si>  
Date: Fri, 01 Sep 95 15:19:31 UTC  
To: hfsig@tapr.org  
Message-ID: <79158@ljutcp.hamradio.si>  
Rendered body type: text/plain

I am amzed by KA9Q FFT results on Pentium as his times are beter than many DSP chips with hardware buterflies! I wonder what kind of performance one can expect for ubiquitous Multiply\_And\_Accumulate filter function with two pointers parallel update? I tried to get this info from many RISC manufacturers to no avail.

I greatly enjoy this conference and this is my first humble posting. I am involved in FFT DSP decoding of HF Morse signals with some good results in 1990

73 de Mario, S56A, N1YU.

---

## 2. Re: [HFSIG:572] FFT v. MAC

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Fri, 1 Sep 1995 10:16:04 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <10413876469@frl.orst.edu>  
Rendered body type: text/plain

Hi Mario,

Welcome again! I am pleased to hear from you. It appears that HFSIG draws from many disciplines - really amazing.

I think Phil is showing that DSP has become possible on the higher-end Intel platforms and, no doubt, we will hear more of this in the future. There is also the P55C to watch - Intel promises that it will be a prospect for "low-end" DSP applications for portable computers, i.e., modems, FAX, and sound etc. I have not seen much on it - it may well have MAC capabilities.

One often see the question coming up about such programs as Baycom running under Linux or Windows and the point being made that it's not viable because of the high demand on the processor. Although this is true if the low-level code is run from a high-end interface, i.e., in a "DOS box". If, however, this code is re-written to run at the lowest ring, and make use of the advanced hardware that is available, it probably will work just fine without too much of a load on the CPU. When working at this level of the OS, the problem is that one have to give up the safety net offered by the OS, i.e. crash and memory protection - in addition, programming at that level is very challenging to say the least, though it could be done. Perhaps that is were our feature challenge lays: doing this with the least amount of hassle. I suspect that it only need to take one good example to show how this is done - anyone ready for a good challenge?

--Johan

---

## 3. Re: [HFSIG:572] FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 1 Sep 1995 18:38:23 -0700  
To: hfsig@tapr.org  
Message-ID: <199509020138.SAA08135@servo.qualcomm.com>  
Rendered body type: text/plain

>I am amzed by KA9Q FFT results on Pentium as his times are beter than many DSP  
>chips with hardware buterflies! I wonder what kind of performance one can

Yup, I keep questioning the need for special purpose processors given the rapidly increasing performance (and declining costs and expanding availability) of general purpose CPUs. But people keep on talking about building TNCs and DSP coprocessors, so I thought I'd convince them with an example. :-)

Last night I went even faster: 333 microseconds (vs yesterday's 380).

Again, this is for a decimation-in-frequency 256-point radix-4 complex

FFT in single-precision (32-bit) floating point. The output is left in bit-reversed order; it can be restored to normal order with a separate routine. Since an important use of the FFT is fast convolution with a fixed impulse response, the DIF FFT is more useful than the decimation-in-time FFT since you can bit-reverse the transform of the impulse response, do your frequency-domain multiplies in bit-reversed order and then do the IFFT while avoiding realtime bit reversals altogether.

The speed of my FFT comes from two radix-4 butterfly routines in assembler. One does a general 4-point butterfly given a starting point, a step between elements, and a step between twiddle factors in a lookup table. The other does a "simple" 4-point butterfly on adjacent operands without twiddles. This is an important special case as the last stage of a DIF consists entirely of them.

The trick is in keeping the Pentium's 3-stage floating point pipeline as full as possible. It really is like juggling with 3 balls in the air at once, being careful to give each ball its allotted time in the air.

Driving these assembler routines are a family of C routines that operate recursively. For example, the 16-point FFT routine calls the general radix-4 butterfly 4 times, then it calls the simple 4-point butterfly four times. The 64-point FFT routine calls the general radix 4 routine 16 times, and then it calls the 16-point routine 4 times on the results, with each 16-point call operating as described above.

This may seem like more overhead than the usual Cooley-Tukey FFT structure with nested loops, but each radix-4 butterfly does enough to make the call/return overhead almost nil. (One of several reasons to do radix-4, btw). It gives a nice elegant structure to the code, but more importantly it changes the memory reference patterns so as to improve the performance of the cache. Franklin Antonio pointed out to me that the normal Cooley-Tukey structure accesses memory sequentially, which is just about pessimal from a cache standpoint. The recursive structure tends to work on one area of memory as long as possible, which greatly improves the cache hit ratio.

This isn't important for small FFTs, but it is important for big ones. Before I updated my asm code last night, I was doing megapoint (1024x1024) CFFTs in about 18.5 seconds. (I haven't timed the new asm code on big arrays yet, but if the speedup is proportional I'd expect about 16 seconds.) This makes it a serious contender for the racks of special purpose hardware the SETI folks have built over the past ten years. Now that Congress has cancelled NASA's SETI funds, maybe this stuff could be useful to the amateur groups who are taking up the effort.

Phil

---

#### 4. Re: [HFSIG:572] FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 1 Sep 1995 18:39:00 -0700  
To: hfsig@tapr.org  
Message-ID: <199509020139.SAA08147@servo.qualcomm.com>  
Rendered body type: text/plain

>expect for ubiquitous Multiply\_And\_Accumulate filter function with two pointers

That's my next target. Again, the trick will be in keeping the Pentium's pipeline full.

Phil

---

#### 5. Re: [HFSIG:573] Re: FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 1 Sep 1995 18:46:01 -0700  
To: hfsig@tapr.org  
Message-ID: <199509020146.SAA08167@servo.qualcomm.com>  
Rendered body type: text/plain

>One often see the question coming up about such programs as Baycom  
>running under Linux or Windows and the point being made that it's not  
>viable because of the high demand on the processor. Although this is true  
>if the low-level code is run from a high-end interface, i.e., in a "DOS  
>box". If, however, this code is re-written to run at the lowest ring, and  
>make use of the advanced hardware that is available, it probably will work

The \*only\* thing that needs to run at "low ring" (or more likely at interrupt time) is a simple FIFO buffering routine that keeps raw data from being lost (on input) or queues from draining (on output) due to scheduler delays. Everything else (software modems, protocols, etc) will run just fine at normal task level.

As long as the low-level buffers are large enough to accomodate the amount of raw data (audio samples, etc) between scheduling ticks of the OS, everything should run quite smoothly.

All this assumes you have enough total CPU cycles to run everything that has to be run. If you don't, then the level at which you run your modem won't make any difference.

Phil

---

## 6. Re: [HFSIG:574] Re: FFT v. MAC

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sat, 2 Sep 1995 17:06:23 +0200  
To: hfsig@tapr.org  
Message-ID: <199509021506.RAA23110@foxbat.pix.za>  
Rendered body type: text/plain

>>  
>Yup, I keep questioning the need for special purpose processors given  
>the rapidly increasing performance (and declining costs and expanding  
>availability) of general purpose CPUs. But people keep on talking  
>about building TNCs and DSP coprocessors, so I thought I'd convince  
>them with an example. :-)  
>

Interesting Phil.

1) I suppose my 486DX2-66 is way to slow for any 'DSP' attemp in realtime ?

2) Why don't you implement a AFSK modem for 1200baud packet on the Pentuim to prove the usefulness of your programs ? It could interface to your NOS s/w...The comport must connect virtually directly to the radio..

73 danie

---

## 7. Re: [HFSIG:577] Re: FFT v. MAC

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sat, 02 Sep 1995 13:07:17 -0800  
To: hfsig@tapr.org  
Message-ID: <9509022004.AA14853@ucs.orst.edu>  
Rendered body type: text/plain

>>>  
>>Yup, I keep questioning the need for special purpose processors given  
>>the rapidly increasing performance (and declining costs and expanding  
>>availability) of general purpose CPUs. But people keep on talking  
>>about building TNCs and DSP coprocessors, so I thought I'd convince  
>>them with an example. :-)  
>>  
>  
>Interesting Phil.  
>  
>1) I suppose my 486DX2-66 is way to slow for any 'DSP' attemp in realtime ?  
>  
>2) Why don't you implement a AFSK modem for 1200baud packet on the Pentuim  
>to prove the usefulness of your programs ? It could interface to your NOS  
>s/w...The comport must connect virtually directly to the radio..  
>  
>73 danie  
>  
>

Wow, I am really impressed Phil - keep up the good work.

Just another two-cents worth: I just have a couple of things that I have been thinking about.

1) I note that you are running BSD, but wonder what you have in mind for the rest of the application, like I/O drivers and the user interface (UI)?

I have managed to port some sound card DSP applications to run on the X-platform under Linux, and I must admit, I really enjoy the real-time

performance in conjunction with the pretty UI. Linux offers a number of mechanisms (three that I am aware of) to do real time work, however, none is very easy. Win95 has similar prospects, though I have not yet gotten that far yet.

2) The price/performance of a high-end processor is very appealing (I was looking at the 133 MHz P5 motherboards - they are now in the order of \$1300). If we look a bit further down the line at the possibility of doing a full radio modem, for example, perhaps some additional considerations beside raw horsepower may be necessary. I am thinking here about timing resources sources - it has to be very reliable and accurate. A relatively simple ARQ protocol like AMTOR or Pactor for example, relies entirely on good timing, i.e., in the order of 30 ppm (CCIR calls for 50 ppm for SITOR). The prospect of raw horsepower, in my opinion, puts the cherry on the cake as then you can freely implement exotic ECC and symbol sync. schemes without running out of steam as often is the case. Obtaining such good timing references off PC platforms is possible, however, though not as simple as one might expect.

I have seen some mention of a version of SPOX for the P55C. That probably will provide all the necessary inter task communications mechanisms, timing, and scheduling you need. And probably co-exist on the same platform as the mainstream GUI OS.

Has anyone heard anything more about these developments besides the vague announcements in the press about Intel's NSP?

--Johan

---

## 8. Re: [HFSIG:577] Re: FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Sat, 2 Sep 1995 23:20:19 -0700  
To: hfsig@tapr.org  
Message-ID: <199509030620.XAA16187@servo.qualcomm.com>  
Rendered body type: text/plain

My pentium does FFTs "only" 100x faster than necessary to handle data sampled at 9600 Hz. I haven't run any tests, but I suspect even a DX2-66 can run with cycles to spare. And you can always pop in a DX4-100 if you like.

>2) Why don't you implement a AFSK modem for 1200baud packet on the Pentuim

Why bother implementing a modem that was obsolete before it was even adopted by amateur packet radio? :-) I'd rather concentrate on something more efficient. Today I prototyped a piccolo modulator. You specify symbol rate, bits/symbol, first tone frequency and tone interval. Uses a Kaiser window with specified beta to shape each symbol; it's interesting to listen to the difference this makes.

The Kaiser window has a parameter, beta, that controls the tradeoff between main lobe width and sidelobe amplitude. beta=0 corresponds to a rectangular window, i.e., no window at all. beta=3.86 is roughly equivalent to a Von Hann window, while beta=7 corresponds to a Blackman window.

BTW, Jim Kaiser and I were in the same group at Bellcore in the mid 1980s. A very nice and helpful guy. I think he retired some time ago.

Phil

---

## 9. Re: [HFSIG:578] Re: FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Sat, 2 Sep 1995 23:25:48 -0700  
To: hfsig@tapr.org  
Message-ID: <199509030625.XAA16195@servo.qualcomm.com>  
Rendered body type: text/plain

Johan,

>1) I note that you are running BSD, but wonder what you have in mind for the >rest of the application, like I/O drivers and the user interface (UI)?

I'm currently using a Soundblaster 16 driver that I hacked into KA9Q NOS a while ago. My BSDI system talks to it over the house ethernet using a TCP connection. I'm doing that for expediency until I find (or write) a Soundblaster 16 driver for BSDI. Or switch to Linux/FreeBSD/NetBSD.

The driver doesn't have to do much -- just buffer raw samples and provide hooks to the app for setting the SB16 operating mode and mixer

settings.

Funny you should mention timing. A bunch of us recently obtained some surplus GPS receivers that we're adapting to PC clock-setting, among other uses. Even without a local GPS receiver, I can set my BSDI system's clock to within a few ms by firing up NTP, the Internet Network Time Protocol.

Phil

---

## 10. Re: [HFSIG:580] Re: FFT v. MAC

From: jmorriso@bogomips.ee.ubc.ca (John Paul Morrison)  
Date: Sun, 3 Sep 1995 11:45:01 -0800 (PDT)  
To: hfsig@tapr.org  
Message-ID: <m0spk2Q-000TsoC@bogomips.ee.ubc.ca>  
Rendered body type: text/plain

>  
> Johan,  
>  
> I'm currently using a Soundblaster 16 driver that I hacked into KA9Q NOS  
> a while ago. My BSDI system talks to it over the house ethernet using  
> a TCP connection. I'm doing that for expediency until I find (or write)  
> a Soundblaster 16 driver for BSDI. Or switch to Linux/FreeBSD/NetBSD.

Linux/FreeBSD/NetBSD share the same sound card drivers (the driver started life under Linux, then it was ported. It appears to have been ported to SCO and UnixWare too). They also share the 387 math emulator as well (unless the BSD guys backed it out political reasons).

Too bad Linux/FreeBSD/NetBSD don't share the same amateur radio networking code. People seem to have gone in different directions in that area.

>  
> Phil  
>

-----  
BogoMIPS Research Labs -- bogosity research & simulation -- VE7JPM --  
jmorriso@bogomips.ee.ubc.ca ve7jpm@ve7jpm.ampr.org jmorriso@ve7ubc.ampr.org  
-----

---

## 11. Re: [HFSIG:579] Re: FFT v. MAC

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sun, 03 Sep 1995 16:49:58 -0800  
To: hfsig@tapr.org  
Message-ID: <9509032346.AA06646@ucs.orst.edu>  
Rendered body type: text/plain

Phil,

> Today I prototyped a piccolo modulator. You  
>specify symbol rate, bits/symbol, first tone frequency and tone  
>interval. Uses a Kaiser window with specified beta to shape each  
>symbol; it's interesting to listen to the difference this makes.  
>  
>The Kaiser window has a parameter, beta, that controls the tradeoff between  
>main lobe width and sidelobe amplitude. beta=0 corresponds to a rectangular  
>window, i.e., no window at all. beta=3.86 is roughly equivalent to a Von Hann  
>window, while beta=7 corresponds to a Blackman window.

Very interesting. A couple of us recently had a private go-around on this topic that proved to be very educational (to me at least). It mainly concerned the rationale behind Dolph-Chebyshev window shapes. I am grateful to Frode for bringing some notes by Harris regarding this topic to my attention. This has to do with how these filter shapes affect dynamic range when using the DFT in multitone systems. It is a bit involved for causal explanations in a few paragraphs, but I am sure the topic will come up in future. As you say, one can hear the difference by ear. How about bringing some of your findings to the DCC and we can compare notes?

>>BTW, Jim Kaiser and I were in the same group at Bellcore in the mid 1980s.  
>A very nice and helpful guy. I think he retired some time ago.

An extraordinary opportunity, Phil. What was the incentive for Kaiser's work? The Dolph-Chebyshev work came about when Dolph was designing multiple antenna arrays and was optimizing main lobe vs. sidelobe patterns of such arrays. Neat eh? Isn't it wonderful how mathematics find it's way into



everything?

--Johan

---

## 12. more Pentium DSP figures

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 5 Sep 1995 02:11:10 -0700  
To: hfsig@tapr.org  
Message-ID: <199509050911.CAA01253@servo.qualcomm.com>  
Rendered body type: text/plain

I'm now down to 318 microseconds on my 256-point CFFT, thanks to some minor tweaking of the complex multiply code sequences.

I came up with a pipelined FIR multiply-accumulate sequence for the Pentium that should execute in 3 clocks/tap once the pipe comes up to speed. I coded up a 16-tap version in assembler along with a C driver to test it. It takes 7.24 seconds to execute the 16-tap FIR routine in a loop 10 million times. That works out to about 22.1 million single-precision multiply-accumulates per second. At 3 clocks/tap on a Pentium 90 that should be 30 million, so if you figure the difference is due to loop overhead that's just about what I expected. Longer filters should asymptotically approach 30 million taps/sec.

Who needs DSPs chips? :-)

Phil

---

## 13. Re: [HFSIG:582] Re: FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 5 Sep 1995 20:09:31 -0700  
To: hfsig@tapr.org  
Message-ID: <199509060309.UAA09280@servo.qualcomm.com>  
Rendered body type: text/plain

Well, I don't know about Dolph and Chebyshev, but Kaiser did his work for the same reason as Hamming, Von Hann, Blackman, et al -- to provide a way to knock down the nasty sidelobes you get from Fourier transforming a finite-time signal with rectangular windowing. You'll usually find a discussion of Kaiser windows along with the others in most DSP textbooks.

The Kaiser window is not as popular as the others probably because it is more complex to evaluate. Among other things, you need the  $i_0()$  function (zeroth-order modified Bessel function of the first kind), which is not in most math routine libraries. But the Kaiser window is the only one I know of with a built-in knob that lets you vary the tradeoff between sidelobe height and main lobe width. You can go all the way from rectangular (i.e., no) window with  $\beta=0$  to something approximating a Blackman window at  $\beta=7$  with a fairly wide ( $\sim 3\times$ ) main lobe but with sidelobes down more than 60dB.

Ordinarily you have to switch between entirely different windows to do this, and you only get a few discrete choices. Kaiser gives you a continuous range.

These windows have lots of uses, including controlling spectral "leakage" in FFTs of continuous sample streams, designing FIR filters (see Rabiner & Gold for a good discussion of using the Kaiser window here), and designing communication signal pulse shapes. I suppose they might even be useful in designing antennas. All these situations involve making inherent tradeoffs between sidelobe amplitudes and mainlobe widths.

Those of you who might have heard of K. Feher's supposedly brilliant and non-obvious (and patented) nonlinear filter for PSK and MSK spectrum control should now recognize it as nothing more than windowing each data pulse with a von Hann window, i.e., a half cosine -- something that's been in the literature for decades. Funny how the same old math keeps getting rediscovered by newcomers. At least Cooley and Tukey didn't try to patent the FFT when they (re)discovered it in the mid 1960s -- it turns out that the algorithm can be traced all the way back to Gauss in the early 1800s!

Phil

---

## 14. Re: [HFSIG:584] Re: FFT v. MAC

From: Alan Bloom <alanb@tsnake2.sr.hp.com>  
Date: Wed, 6 Sep 1995 14:30:39 -0800 (PDT)  
To: hfsig@tapr.org (Phil Karn)  
Message-ID: <199509062130.AA190673040@tsnake2.sr.hp.com>  
Rendered body type: text/plain

Phil wrote:

>  
> These windows have lots of uses, including controlling spectral  
> "leakage" in FFTs of continuous sample streams, designing FIR filters  
> (see Rabiner & Gold for a good discussion of using the Kaiser window  
> here), and designing communication signal pulse shapes.

I've been playing around with MathCad to see how big an FIR filter I need to implement a root raised cosine filter, such as is used to reduce the transmitted bandwidth of many digital modulation modes. With an  $\alpha=0.15$  filter, simply truncating the impulse response at 16 symbols wide results in about -33 dB sidelobes.

Now I'm about to try different windowing functions. Before I re-invent the wheel, can anyone just tell me what's the best window to use for minimum sidelobes? What I care about is the worst-case number. Reducing far-out sidelobes doesn't do me any good if a closer one is higher.

AL N1AL

---

## 15. Re: [HFSIG:585] Re: FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Wed, 6 Sep 1995 19:19:39 -0700  
To: hfsig@tapr.org  
Message-ID: <199509070219.TAA20272@servo.qualcomm.com>  
Rendered body type: text/plain

There is no fixed answer to your question. You also need to consider how much passband ripple and how wide a passband/stopband transition region you're willing to tolerate.

The Remez exchange algorithm seems to be the preferred way to design FIR filters, mainly because it's readily adaptable to arbitrary shapes and can add controlled ripple.

Phil

---

## 16. Re: [HFSIG:585] Re: FFT v. MAC

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Wed, 06 Sep 1995 20:08:05 -0800  
To: hfsig@tapr.org  
Message-ID: <9509070304.AA16870@ucs.orst.edu>  
Rendered body type: text/plain

Al,

>Phil wrote:  
>>  
>> These windows have lots of uses, including controlling spectral  
>> "leakage" in FFTs of continuous sample streams, designing FIR filters  
>> (see Rabiner & Gold for a good discussion of using the Kaiser window  
>> here), and designing communication signal pulse shapes.  
>  
>I've been playing around with MathCad to see how big an FIR filter I  
>need to implement a root raised cosine filter, such as is used to  
>reduce the transmitted bandwidth of many digital modulation modes.  
>With an  $\alpha=0.15$  filter, simply truncating the impulse response  
>at 16 symbols wide results in about -33 dB sidelobes.  
>  
>Now I'm about to try different windowing functions. Before I re-invent  
>the wheel, can anyone just tell me what's the best window to use for  
>minimum sidelobes? What I care about is the worst-case number. Reducing  
>far-out sidelobes doesn't do me any good if a closer one is higher.  
>  
>AL N1AL  
>  
>

You may want to look at Fred Taylor's book: Digital Filter Design Handbook. Dekker, ISBN 0-8247-1357-5. Chapter 5 is devoted to this topic - Pages 150,151 shows a number of windows and how they trade off characteristics. He also shows how to compute the Kaiser window.

Another excellent source of information that will definately be of interest to you, is H.D. Helm's paper: Digital Filters with Equiripple or Minimax responses. IEEE Trans. on Audio and Electroacustics. VOL. AU-19, No.1 March 1971. p:87-93 - this will lead you to the Dolph-Chebyshev work. Essentially lets you specify the frequency domain response then perform an IFFT followed by Dolph-Chebyshev smoothing, to balance the response for time domain filter coefficients. Only a couple of lines of Matlab.

Hope this helps.

--Johan

---

## 17. Re: [HFSIG:583] more Pentium DSP figures

From: Marijan Miletic <s56a@ljutcp.hamradio.si>  
Date: Thu, 07 Sep 95 06:47:52 UTC  
To: hfsig@tapr.org  
Message-ID: <80535@ljutcp.hamradio.si>  
Rendered body type: text/plain

Hi HF-SIG's,

Thanks for a kind support of my first posting in this conference.

Once again, I am amazed by KA9Q MAC figures. Even with more taps, I am sure all the coefficients and circular data buffer might be kept in cache for quick pipeline filling. MAC assembler loop is short enough to fit separate instruction pipeline and out comes 30 MTS!

Although Pentium floating point processor is 3 times faster than 486, I feel lot of useful audio range DSP can be done even by my old 486/33.

>Who needs DSPs chips? :-)

They seem to be good bit fidlers!

73 de Mario, S56A, N1YU.

---

## 18. Re: [HFSIG:585] Re: FFT v. MAC

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)  
Date: Thu, 7 Sep 95 07:39:57 CDT  
To: hfsig@tapr.org  
Message-ID: <9509071239.AA01752@eagle.aud.alcatel.com>  
Rendered body type: text/plain

> Phil wrote:  
> >  
> > These windows have lots of uses, including controlling spectral  
> > "leakage" in FFTs of continuous sample streams, designing FIR filters  
> > (see Rabiner & Gold for a good discussion of using the Kaiser window  
> > here), and designing communication signal pulse shapes.  
>  
> I've been playing around with MathCad to see how big an FIR filter I  
> need to implement a root raised cosine filter, such as is used to  
> reduce the transmitted bandwidth of many digital modulation modes.  
> With an alpha=0.15 filter, simply truncating the impulse response  
> at 16 symbols wide results in about -33 dB sidelobes.  
>  
> Now I'm about to try different windowing functions. Before I re-invent  
> the wheel, can anyone just tell me what's the best window to use for  
> minimum sidelobes? What I care about is the worst-case number. Reducing  
> far-out sidelobes doesn't do me any good if a closer one is higher.  
>  
> AL N1AL  
>  
>

Al: I have implemented compensated- and uncompensated root-cosine FIR filters with excellent sidelobe suppression with 31 taps, using a version of the Remez-exchange code that I have modified for various transition band shapes. Commercially I have seen it done with as few as 24 taps. Windowing functions, such as Hamming and Hanning windows can give good out-of-band suppression, but at the expense of transition-band distortion, which results in eye-pattern closure, in my case, about 1 - 2 dB.

I have written a book that TAPR will publish this fall, "Handbook of RF Modem Theory and Design" which covers this subject in some detail, and I include a disk with my modified Remez-exchange software. The modifications consist of adding raised cosine, root raised-cosine, and sinc()-compensated root

raised cosine filter shapes. I have plots of frequency response and eye pattern for several cases of impulse response windowing. I also include Excel 5.0 spreadsheets for generating eye pattern from impulse/frequency response.

These were done for alpha's in the range 0.4 to 0.8. At an alpha of 0.15, the filter may have to be longer, but this may not be real practical since low-alpha filters require more careful frequency response shaping, including other factors, such as channel response, IF filter response, and any other distortions, such as AF channel response.

- Tom, N5EG

```
-----+-----
Tom McDermott | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com | represent those of Alcatel
[ICC'96 Technical Program Secretary] | Network Systems, Inc."
[June 23-27, 1996, Dallas, Tx.] |
-----+-----
```

---

## 19. Re: [HFSIG:589] Re: FFT v. MAC

From: Alan Bloom <alanb@tsnake2.sr.hp.com>  
Date: Thu, 7 Sep 1995 11:54:37 -0800 (PDT)  
To: hfsig@tapr.org (Tom McDermott)  
Message-ID: <199509071854.AA118020078@tsnake2.sr.hp.com>  
Rendered body type: text/plain

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom McDermott)  
From: forrerj@ucs.orst.edu (Johan Forrer)  
From: Phil Karn <karn@qualcomm.com>

Thanks Tom, Johan and Phil. Yesterday I tried windowing the 0.15 alpha root-raised-cosine filter's impulse response with a simple Hanning window (16 symbol width stretched by the window to 32) and got -58 dB worst-case stopband ripple. Worst-case error in the frequency response was 0.25 dB up to 1.1 Fn.

That seems pretty good for a first try. A window with a little narrower main lobe in the frequency response should give better pass band accuracy. Perhaps a Dolph-Chebyshev, or a Kaiser (if I can find the reference for Bessel functions!)

Thanks again,

Alan

---

## 20. Re: [HFSIG:590] Re: FFT v. MAC

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom McDermott)  
Date: Thu, 7 Sep 95 14:30:36 CDT  
To: hfsig@tapr.org  
Message-ID: <9509071930.AA01936@eagle.aud.alcatel.com>  
Rendered body type: text/plain

> Thanks Tom, Johan and Phil. Yesterday I tried windowing the 0.15 alpha  
> root-raised-cosine filter's impulse response with a simple Hanning  
> window (16 symbol width stretched by the window to 32) and got -58 dB  
> worst-case stopband ripple. Worst-case error in the frequency response  
> was 0.25 dB up to 1.1 Fn.

>  
> That seems pretty good for a first try. A window with a little narrower  
> main lobe in the frequency response should give better pass band  
> accuracy. Perhaps a Dolph-Chebyshev, or a Kaiser (if I can find  
> the reference for Bessel functions!)

>  
> Thanks again,

>  
> Alan  
>

Alan: you may find that the eye closure and the maximum error in the transition band are not the same, once you hit about the -20 dB point in the response, there is not enough energy left to close the eye very much, so the error is more important (in terms of eye closure) at frequencies lower than that point.

For 2-level eyes it usually is not much of a problem. For

multiple-level eyes, it is quite important.

- Tom, N5EG

```
-----+-----
Tom McDermott | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com | represent those of Alcatel
[ICC'96 Technical Program Secretary] | Network Systems, Inc."
[June 23-27, 1996, Dallas, Tx.] |
-----+-----
```

---

## 21. Re: [HFSIG:583] more Pentium DSP figures

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Fri, 8 Sep 1995 06:21:12 +0200  
To: hfsig@tapr.org  
Message-ID: <199509080421.GAA26506@foxbat.pix.za>  
Rendered body type: text/plain

>  
>Who needs DSPs chips? :-)  
>  
Phil,

However...I need still to be convinced that the CPU can then do anything else but the FFT or FIR ? I mean other house keeping would normally be done in C or so. Or would it also be coded into ASM ?

I guess my point is: who will do the housekeeping and what effect will it have on the Pentium doing its DSP stuff now fulltime say for a modem implementation ?

Just asking, I am no boffin. Just following your work with interest.

73 danie zs6awk

---

## 22. Re: [HFSIG:592] Re: more Pentium DSP figures

From: Phil Karn <karn@qualcomm.com>  
Date: Fri, 8 Sep 1995 18:49:07 -0700  
To: hfsig@tapr.org  
Message-ID: <199509090149.SAA21011@servo.qualcomm.com>  
Rendered body type: text/plain

>However...I need still to be convinced that the CPU can then do anything  
>else but the FFT or FIR ? I mean other house keeping would normally be done  
>in C or so. Or would it also be coded into ASM ?

Well, let's say you needed to do one 256-pt CFFT on each 512 samples of real data that arrive at a 9600 sample/sec rate. That's one FFT every 53.3 milliseconds. If it takes me 320 us to do one FFT, that's 0.6% of the machine's real time. I think I'll have time left to do other things.

Obviously a real modem will have to do more than just a CFFT, but I suspect even after I add the rest of it I'll have plenty of time left.

Most of the code can be in C. Only a few compute-intensive primitives really need to be written in asm language, and even there I don't think it's really essential -- it's just neat to see how fast you can make them go.

Phil

---

## 23. Re: [HFSIG:580] Re: FFT v. MAC

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Sat, 09 Sep 1995 08:47:00 +0100  
To: hfsig@tapr.org  
Message-ID: <199509090852.AA20851@felix.dircon.co.uk>  
Rendered body type: text/plain

> I can set my BSDI  
>system's clock to within a few ms by firing up NTP, the Internet  
>Network Time Protocol.  
>  
>Phil  
>  
>  
Phil what RFC is NTP described in ?

Regards Charles G4GU0

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 24. Re: [HFSIG:594] Re: FFT v. MAC

From: Phil Karn <karn@qualcomm.com>  
Date: Sat, 9 Sep 1995 22:04:30 -0700  
To: hfsig@tapr.org  
Message-ID: <199509100504.WAA28725@servo.qualcomm.com>  
Rendered body type: text/plain

>Phil what RFC is NTP described in ?

1305.

Phil

---

## 25. RE: [HFSIG:594] Re: FFT v. MAC

From: "Beaudry, Tom" <Tom\_Beaudry@tvratings.com>  
Date: Mon, 11 Sep 95 10:24:00 PDT  
To: hfsig <hfsig@tapr.org>  
Message-ID: <30547005@mailgate.dun.nielsen.com>  
Rendered body type: text/plain

Check the US Naval Observatory's home page for a link to the information  
as well as instructions on how to sync your computer's clock to their's.  
The URL is:

<<http://www.usno.navy.mil/home.html>>

-----  
From: hfsig[SMTP:hfsig@tapr.org]  
Sent: Saturday, September 09, 1995 3:55 AM  
To: hfsig  
Subject: [HFSIG:594] Re: FFT v. MAC

> I can set my BSDI  
>system's clock to within a few ms by firing up NTP, the Internet  
>Network Time Protocol.

>  
>Phil  
>  
>

Phil what RFC is NTP described in ?

Regards Charles G4GU0

-----  
"projects don't slip, they just catch up with reality"

Charles Brain (G4GU0)  
Chelmsford, Essex.  
E-mail chbrain@dircon.co.uk  
POTS +44 (0)1245 353221  
FAX +44 (0)1245 275448  
-----

---

## 26. HFSIG at DCC

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Mon, 11 Sep 1995 11:14:11 -0800  
To: HFSIG@TAPR.org  
Message-ID: <9509111810.AA13777@ucs.orst.edu>  
Rendered body type: text/plain

Hi all,

I got back from the DCC late last night - lots of exciting bits to share.

Things are very hectic here at the moment, but I promise to fill you in  
about the main attactions. However, regarding HFSIG's participation,  
congratulations to Pawel on the 15-tone modem - it attracted a lot of

attention, also Adrian's DSP audio processor was of much interest.

A special word of thanks to Walt for bringing his computer that we ran the demo's on.

I much appreciated the discussions during the HFSIG discussion - we have made progress on some rather difficult issues.

May I just add that I appreciated the opportunity being at the DCC and sincerely thank the organisers, especially TPRS, TAPR, and Greg for hosting us and making it a worthwhile experience.

73's

--Johan

---

## 27. [HFSIG:596] Re: FFT v. MAC

From: "Garrett, John, GARRETJ2" <GARRETJ2@btlip15.bt.co.uk>  
Date: Tue, 12 Sep 95 08:19:00 BST  
To: HFSIG@TAPR.ORG  
Message-ID: <30553547@smtpgate.comnet.bt.co.uk>  
Rendered body type: text/plain

Away until 25th September. Please contact Lynn Dunnett on 01473 644859 for assistance in my absence.

---

## 28. OFDM equalization

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)  
Date: Tue, 12 Sep 95 07:55:33 CDT  
To: hfsig@tapr.org  
Message-ID: <9509121255.AA01669@eagle.aud.alcatel.com>  
Rendered body type: text/plain

At the ARRL DCC in Arlington, Texas this weekend, several of us had an interesting conversation regarding the 15-carrier QPSK OFDM system being designed by Pawel. I had the opportunity to see the demonstration, and congratulations are due to Pawel, and crew for some very good work. Johan asked if I could describe our conversation.

During the discussion, the topic of equalization came up, and it was noted that because of the different frequency responses of the equipment, that the tones did not arrive at the receiver with the same amplitude, and this suggested that it might be possible to equalize the received signal amplitude within the DSP receive algorithm.

In the initial transmission phase, I beleive that 4 tones are sent so that synchronization can be achieved. It seems that if the tones are properly spaced in the passband, then simultaneous with the synchronization, the the amplitude of the 4 tones could be measured, and then a simple minimal-pole equalizer formulated to compensate for non-flat response. Perhaps this might require 5 tones, however.

On later reflection, I have the following concern, and possible solution:

While this would equalize the receiver variation, and perhaps make the FFT that converts the time-waveform to the individual amplitude bins somewhat better, if the amplitude distortion is primarily occuring at the transmitter, then it does not really solve the problem of some of the 15 carriers having poorer signal-to-noise ratio on the HF band than other carriers. There is a theorem called the water-pouring-principle which states that the best  $E_b/N_0$  performance occurs when the transmitter power is equally divided between the carriers.

So, it seems that possibly an off-line equalization of the transmitter might be needed, sort of a utility program that let the user record the output power of each of the 15 carriers, and then tell the software about those levels. Next, some compensation for that could be made, and then an amplitude equalizer programmed (once). This equalizer would then be inserted into the transmit side of the modem code, resulting in a particular user's transmitter being more or less optimally flat versus carrier frequency.

Now, at the receive side, an adaptive equalization could occur during the synchronization time period on receive, and computed, then inserted into the receive modem side. This would compensate for any receiver non-flatness, and any left-over equalization errors, resulting in fewer spill-over effects in the receive side FFT. Of course there would have to be care taken that the receiver did not attempt to

equalize during a selective-fading event, otherwise it might compute a poor choice for that frame. But this effect should be corrected next frame, or perhaps could be averaged over several frames.

- Tom, N5EG

```
-----+-----
Tom McDermott | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com | represent those of Alcatel
[ICC'96 Technical Program Secretary] | Network Systems, Inc."
[June 23-27, 1996, Dallas, Tx.] |
-----+-----
```

---

## 29. [HFSIG:598] Re: FFT v. MAC

From: "Garrett, John, GARRETJ2" <GARRETJ2@btlip15.bt.co.uk>  
Date: Wed, 13 Sep 95 08:19:00 BST  
To: HFSIG@TAPR.ORG  
Message-ID: <30568695@smtpgate.comnet.bt.co.uk>  
Rendered body type: text/plain

Away until 25th September. Please contact Lynn Dunnett on 01473 644859 for assistance in my absence.

---

## 30. HFSIG at DCC

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 15 Sep 1995 13:44:06 -0800  
To: HFSIG@tapr.org  
Message-ID: <9509152040.AA24197@ucs.orst.edu>  
Rendered body type: text/plain

Hi,

Sorry for taking a while to get a brief summary out - had to take care of important business. Here is the promised summary of the DCC. Enjoy!

The DCC this year was buzzing with activities. Two outstanding workshops, excellent papers, and interesting talks on amateur radio networking infrastructure developments, future satellites, and other digital issues made it a busy time. In addition, meeting old friends and the opportunity to make new ones is always enjoyable. Not to mention the informal discussions over a beer late at night, listening to exciting things taking place. Unfortunately, a significant proportion of happenings never make it into the proceedings, so it made the extra effort to be there all that much more worthwhile.

Saturday's program consisted of several parallel sessions. It was hard to decide which one to participate in as often the case may be - in my case I was involved in presenting material as well. So keep in mind that I can only relate to what I could fit in these busy few hours. The organizers had an excellent idea to make room for series of introductory presentations: digital communications in general, HF digital, digital satellite communications, and packet networking. Besides giving beginners an opportunity to pick up a wealth of ideas, this also gave local amateurs an opportunity to hear what digital communications were about.

After presenting my introductory talk on HF digital, I slipped out to hear the end of Frank Perkins' presentation on software development issues for the DSP-93. I don't know how many really appreciated how successful the DSP-93 modems are - this is due to the outstanding efforts of the DSP-93 software developers such as Frank. One very useful tip that I picked up from Frank's talk was the problem that the "ABS" function introduce when used in detectors. The "SQUARED" function is what one need, this is because the absolute value of a band-limited signal turns out to be not band-limited and subsequently cannot be lowpass filtered. In addition Frank had a video tape with sound that made for a great presentation. Tom McDermott's talk followed on using Excel for evaluating DSP solutions. It was obvious that Tom has uncovered a powerful tool for serious DSP work. Using built-in Excel functions besides several routines that he developed himself, the point-and-click, drag-and-drop metaphors makes quick work of analyzing sampled signals. When Tom's book becomes available, we will have further opportunities to try these ourselves. Looks like a reasonably inexpensively alternative to



other programs.

The "HF issues" session and Phase3D talk unfortunately ran parallel - we missed a couple of good participants, and I would have loved hearing Bdale Garbee's talk. Anyway, we had excellent participation on important HF digital issues. I gave brief overviews on the status of the HF channel simulator, Adrians's MFSK, and Pawel's OFDM modems. The HF ionospheric simulator work appears to be of some interest and a few moments was spent on that. Paul Rinaldo, W4RI, brought up an interesting idea - how to evaluate adjacent channel interference. The Watterson model, for example, only deals with the variations due to the ionosphere. It would be interesting to have some means for evaluating different situations due to other nearby digital transmissions. Phil Karn's progress on Pentium-powered DSP was noteworthy and stirred interest - that would be an interesting one to watch in future. This was followed by a lively discussion on FCC regulations and the future of using our proposed multi-tone HF modems. It appears that it would be possible to obtain either STA or experimental licenses to experiment. Paul kindly offered to help looking into this.

What helped promote our ideas a bit, was the live demo that we put on afterwards. Walt, K5YFW, brought his computer that we set up to give folks the opportunity to see Pawel's 15-carrier modem in operation. Some mentioned that the audio has a striking likeness to 300 baud HF packet. Many would be fooled into thinking that it was indeed packet - this is because it does not sound like a wideband signal at all. I suspect that a demo such as this may win many over. We also played with the DSP sound card and played with Adrian's, G4ZHZ, signal processing software. I think he did a very nice job to package several really nice functions, i.e., several CW filters and others for SSB, RTTY, and FAX. This also includes a Hilbert transform with which one could slide the output audio pitch to wherever audio frequency you wanted, kind of like a digital BFO. I played some audio from a cassette tape and the ability to shift audio in this manner without clicks, pops or other artifacts, is really amazing - all made possible by DSP.

There was not very much equipment on display except for Paccom that showed a diverse range amateur digital equipment. I did however, took a bit of time looking over the new Pactor II hardware - the hardware is very well made, and the technology very advanced. If you are interested in reading a bit more about the technical aspects of this hardware also about some of software tricks that makes up Pactor II, be sure to ask the folks at Paccom for a data sheet - its very well written.

I left the best for last - Phil Karn's Saturday evening's dinner presentation. In essence, the theme was what the future holds for amateur radio, not only the digital modes, but how to best utilize our portion of the RF spectrum. A large portion of the presentation was devoted to a walk through Phil's Internet URL on coding. This is a remarkably well done, common-sense audio/visual overview of the mathematical magic of some important issues regarding future digital transmission of data and voice: bandwidth, power, information theory, and most important, the mindset of the average amateur. Phil did an outstanding job entertaining us with his vision, and all this with his usual good humor.

Although I had a lot to do on my way home (like working on a business plan), nevertheless I could not help but think about the impressions that the conference left: the contrast between wide band vs. narrow band. I have no doubts in my mind about the technological advantages that wide band offers, but what make me most excited is to think of all the opportunities that lays ahead in digital RF technology and DSP. Would'nt you you agree that this is something to look forward to?

I would like to thanks all the folks that made the conference a success: the ARRL and our hosts APRS and TAPR. Well done and much thanks. Also to the software developers that helped make the demo: Pawel, and the Finn's, Adrian and Walt for bringing his computer equipment. Well done.

Let me know if I missed anything.

73's

--Johan, KC7WW

---

## 31. To: Peter TY1PS

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sat, 16 Sep 1995 14:48:21 -0800  
To: HFSIG@tapr.org  
Message-ID: <9509162144.AA28462@ucs.orst.edu>  
Rendered body type: text/plain

Hi Peter,

I have somehow managed to loose your e-mail address. Could you please be so kind and e-mail me - I have some further news that might interest you.

73's

--Johan

---

## 32. Re: [HFSIG:601] HFSIG at DCC

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sun, 17 Sep 1995 09:41:41 +0200  
To: hfsig@tapr.org  
Message-ID: <199509170741.JAA03183@foxbat.pix.za>  
Rendered body type: text/plain

>Hi,  
>  
>Sorry for taking a while to get a brief summary out - had to take care of  
>important business. Here is the promised summary of the DCC. Enjoy!  
>  
>  
>  
> The DCC this year was buzzing with activities. Two....

Dankie Johan vir die terugvoer. Jy moet my laat weet as enige van die 'talks' op Internet gaan verskyn of bestel kan word per pos. Dit lyk of RF digital kommunikasie baie 'exciting' tye binnegaan. Ek is ook opgewonde. Daar is meer en meer spread spectrum goed aan die gang sien ek. Jy weet seker dat mens nogal goedkoop sprei spektrum chippies kan koop deesdae. Ken jy bv die STEL-2000A, die S20043 van AMI, die magtige UNISYS chip ens ens Van hulle kos selfs hier in ZS so laag as R160 ! Lokaal is daar egter nie belangstelling in sulke gevorderde tegnologie nie hi !

Ek het vanoggend weer CW op 40m gewerk na 'n lamg tyd. Ek werk toe 'n ZS2lm en hy se 'aangename kennis, is jy nuut op die band ?' ek se toe nee geen my net kans my CW sal reg kom hi ! (ons het so 22wpm gewerk)

groete

Danie ZS6AWK

---

## 33. Re: [HFSIG:603] Re: HFSIG at DCC

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sun, 17 Sep 1995 18:27:50 -0800  
To: hfsig@tapr.org  
Message-ID: <9509180123.AA15572@ucs.orst.edu>  
Rendered body type: text/plain

>>Hi,  
>>  
>>Sorry for taking a while to get a brief summary out - had to take care of  
>>important business. Here is the promised summary of the DCC. Enjoy!  
>>  
>>  
>>  
>> The DCC this year was buzzing with activities. Two....  
>  
>Dankie Johan vir die terugvoer. Jy moet my laat weet as enige van die  
>'talks' op Internet gaan verskyn of bestel kan word per pos.

Die "proceedings" is beskikbaar vanaf ARRL (14th DCC). Daar is egter baie min inhoud wat werklik daar aangegaan het. Meeste van die artikels in die "proccedings" was nie eens voorgedra nie - net ingestuur vir die konverensie. Andersins weet ek nie van enige van die praatjies wat elektronies beskikbaar gaan wees nie, behalwe Phil se URL op Internet WWW.

>Dit lyk of RF  
>digital kommunikasie baie 'exciting' tye binnegaan. Ek is ook opgewonde.  
>Daar is meer en meer spread spectrum goed aan die gang sien ek. Jy weet  
>seker dat mens nogal goedkoop sprei spektrum chippies kan koop deesdae. Ken  
>jy bv die STEL-2000A, die S20043 van AMI, die magtige UNISYS chip ens ens  
>Van hulle kos selfs hier in ZS so laag as R160 ! Lokaal is daar egter nie

>belangstelling in sulke gevorderde tegnoligie nie hi !

Daar was heelwat tyd aan die chips bestee in privaat geselskap. Sal die ontwikkelings fyn dophou.

>  
>Ek het vanoggend weer CW op 40m gewerk na 'n lamg tyd. Ek werk toe 'n ZS21m  
>en hy se 'aangename kennis, is jy nuut op die band ?' ek se toe nee geen my  
>net kans my CW sal reg kom hi ! (ons het so 22wpm gewerk)  
>  
>groete  
>  
>Danie ZS6AWK  
>  
>

Wou nog iets se i.s. jou vraag oor die EVM en RTTY, maar vroulied het my aangejaag dat ons moes vertrek op pad kerk toe.

Hoe genruik jy jou INT's op die com poorte? Is jy seker dat dit uniek is, maw is daar nie 'n moontlikheid dat daar "device" drivers die interrupts onderskep nie? Die lyk amper asof jy 'n mouse driver het wat die serie poort data onderskep en dink jy beweeg die muis? Jy moet moontlik die mouse drywer uithaal uit jou autoexec.?

Nog 'n idee: hoe skakel jy die poorte om van DEBUG na SCI? Twee kables na die PC, of een kabel van die PC wat jy omruil op die EVM?

Hoop dit gee 'n paar idees oor jou probleem, want anders vermoed ek die ergste - 'n probleem met jou DSP chip. Hoop nie dis die geval nie.

Groete tot later,

--Johan

---

### 34. [HFSIG:601] HFSIG at DCC

From: "Garrett, John, GARRETJ2" <GARRETJ2@btlip15.bt.co.uk>  
Date: Mon, 18 Sep 95 08:18:00 BST  
To: HFSIG@TAPR.ORG  
Message-ID: <305D1E1F@smtpgate.comnet.bt.co.uk>  
Rendered body type: text/plain

Away until 25th September. Please contact Lynn Dunnett on 01473 644859 for assistance in my absence.

---

### 35. Pitch shifting

From: "Alberto di Bene (xx39-2-596.25744)" <dibene@VNET.IBM.COM>  
Date: Mon, 18 Sep 95 12:42:51 GMT  
To: hfsig@tapr.org  
Message-ID: <199509181047.FAA13795@sys1.tapr.org>  
Rendered body type: text/plain

> This also includes a Hilbert transform with which one could  
> slide the output audio pitch to wherever audio frequency you  
> wanted, kind of like a digital BFO. I played some audio from a  
> cassette tape and the ability to shift audio in this manner  
> without clicks, pops or other artifacts, is really amazing - all  
> made possible by DSP.

Do you mean that the audio band is shifted (keeping the bandwidth), or that it is multiplied by some factor ?

Do you have more info on this subject ? I have a DSP evaluation board with the TMS320C50 and would like to try to implement it.

Thanks for any info

73 de  
Alberto di Bene, I2PHD i2phd@amsat.org diben@vnet.ibm.com

---

### 36. Re: [HFSIG:606] Pitch shifting

From: Phil Karn <karn@qualcomm.com>  
Date: Mon, 18 Sep 1995 10:37:56 -0700  
To: hfsig@tapr.org  
Message-ID: <199509181737.KAA26088@servo.qualcomm.com>  
Rendered body type: text/plain

> Do you mean that the audio band is shifted (keeping the bandwidth), or  
> that it is multiplied by some factor ?

Shifted, keeping the bandwidth -- just like a SSB receiver. That's why the Hilbert Transform is required.

If you compose a complex signal from a real-valued signal as the real part and the hilbert transform of the real part as the imaginary part, then the fourier transform of the complex signal is one-sided. That means you can shift it in frequency (e.g., by multiplying by  $e^{j\omega t}$ ) without any pesky negative frequencies aliasing around zero back into the audio band.

This is the basis of a phasing-type SSB exciter -- you do the  $*e^{j\omega t}$  part by running the complex signal into an I/Q modulator and summing the results.

Phasing-type SSB exciters fell by the wayside when crystal filters did the job, because they required no careful adjustments. I predict phasing will make a comeback now that you can do the job in DSP much better.

Phil

---

### 37. FCC regulations and multitone modems

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Wed, 20 Sep 1995 08:31:57 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <2C1C4B364A@frl.orst.edu>  
Rendered body type: text/plain

Hi all,

I did mention that we discussed the legal status of US amateur radio operator regarding the use of multitone modems on HF, at the recent DCC.

I just heard from Chris Imlay, N3AKD in Washinton that he and Paul Rinaldo, the ARRL's liason with the FCC, checked the regulations and found no reason why we could not use our multitone modems on HF, except for unattended operations, where the restriction is now 500 Hz.

This is good news. I take it from this good authority then that we can conduct our experiments on an international basis, something that would have been a problem with an STA.

So, anyone interested in doing local trials for starters? All that is needed is a 386/486 computer running NOS, the Motorola EVM, a rather simple PTT interface, and appropriate cable to connect your transceiver to the EVM.

73's

--Johan, KC7WW

---

### 38. Re: [HFSIG:608] FCC regulations and multitone modems

From: Phil Karn <karn@qualcomm.com>  
Date: Wed, 20 Sep 1995 17:10:40 -0700  
To: hfsig@tapr.org  
Message-ID: <199509210010.RAA29966@servo.qualcomm.com>  
Rendered body type: text/plain

Johan,

I'm confused. Aren't there both subband and bandwidth problems on HF? As I understood it, data is not allowed in the voice bands (except for digital voice), and down in the CW bands I didn't know we could run data with ~3 KHz bandwidths. Have the rules changed recently? I admit to not having kept up.

Phil

---

### 39. Re: [HFSIG:604] Re: HFSIG at DCC

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Thu, 21 Sep 1995 06:55:13 +0200  
To: hfsig@tapr.org  
Message-ID: <199509210455.GAA19796@foxbat.pix.za>  
Rendered body type: text/plain

Johan

>Hoe genruik jy jou INT's op die com poorte? Is jy seker dat dit uniek is,  
>maw is daar nie 'n moontlikheid dat daar "device" drivers die interrupts  
>onderskep nie? Die lyk amper asof jy 'n mouse driver het wat die serie poort  
>data onderskep en dink jy beweeg die muis? Jy moet moontlik die mouse drywer  
>uithaal uit jou autoexec.?

Ja jy is miskien reg ! Ek het vergeet van die device drivers wat daar sit en  
die int vektore verander, sal jou laat weet...

>Nog 'n idee: hoe skakel jy die poorte om van DEBUG na SCI? Twee kables na  
>die PC, of een kabel van die PC wat jy omruil op die EVM?

Ek het twee kables met db9's aan na com1 en com2. My debugger is op com2 en  
die SCI op com1.

73 Danie

PS O ja wou jou nog se; hulle is besig om 'n 64kbps lyn na ons mpy in te  
sit. Op die stadiuim het ek reeds Netcape @ 64kbps. Dis fantasties en wil nou  
bietjie ou Phil se .wav ens demo's aflaai op 'n dag. EK kan egter nog nie  
FTP en email nie. Dis nog nie 'aangeskakel' nie.

---

#### 40. Re: [HFSIG:610] Re: HFSIG at DCC

From: Greg Jones <gjones@tenet.edu>  
Date: Thu, 21 Sep 1995 00:35:05 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199509210535.AAA08720@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

For us Americans on-line, can someone translate :-)

You know the joke, don't you ?

What do you call someone who speaks three languages -- tri-lingual  
and someone who speaks two languages -- bi-lingual

and someone who speaks one language ?

Another dran American tourist.

Cheers - greg

According to Danie Brynard:

>  
> Johan  
>  
> >Hoe genruik jy jou INT's op die com poorte? Is jy seker dat dit uniek is,  
> >maw is daar nie 'n moontlikheid dat daar "device" drivers die interrupts  
> >onderskep nie? Die lyk amper asof jy 'n mouse driver het wat die serie poort  
> >data onderskep en dink jy beweeg die muis? Jy moet moontlik die mouse drywer  
> >uithaal uit jou autoexec.?  
>  
>  
> Ja jy is miskien reg ! Ek het vergeet van die device drivers wat daar sit en  
> die int vektore verander, sal jou laat weet...  
>  
>  
> >Nog 'n idee: hoe skakel jy die poorte om van DEBUG na SCI? Twee kables na  
> >die PC, of een kabel van die PC wat jy omruil op die EVM?  
>  
> Ek het twee kables met db9's aan na com1 en com2. My debugger is op com2 en  
> die SCI op com1.  
>  
> 73 Danie  
>  
> PS O ja wou jou nog se; hulle is besig om 'n 64kbps lyn na ons mpy in te  
> sit. Op die stadiuim het ek reeds Netcape @ 64kbps. Dis fantasties en wil nou  
> bietjie ou Phil se .wav ens demo's aflaai op 'n dag. EK kan egter nog nie  
> FTP en email nie. Dis nog nie 'aangeskakel' nie.  
>  
>

---

#### 41. Re: [HFSIG:607] Re: Pitch shifting

From: dibene@VNET.IBM.COM (Alberto di Bene)  
Date: Thu, 21 Sep 95 16:20:15 GMT  
To: hfsig@tapr.org  
Cc: KARN@QUALCOMM.COM  
Message-ID: <199509211458.JAA30046@sys1.tapr.org>  
Rendered body type: text/plain

In a previous message, you wrote:

> ...  
> Phasing-type SSB exciters fell by the wayside when crystal filters did  
> the job, because they required no careful adjustments. I predict phasing  
> will make a comeback now that you can do the job in DSP much better.  
> ...

Thanks for the explanation Phil. Your prediction seems to be already true. Have you seen the new ICOM 775-DSP ? They claim SSB generation by mean of a DSP-controlled PSN (Phase Shift Network?)

Alberto di Bene                dibene@vnet.ibm.com            i2phd@amsat.org

-----  
Purified Air Environment. Please do not open Windows.                )  
-----

---

## 42. Re: [HFSIG:609] Re: FCC regulations and multitone modems

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Thu, 21 Sep 1995 08:46:22 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <445A6D254D@frl.orst.edu>  
Rendered body type: text/plain

Hi Phil,

Although I admit that I have not seen the very latest regs, I assume that the rules/subbands for data transmission apply. The new regs apparently has 500Hz bandwidth for unattended data transmission, also fully legalize Clover II and Pactor I (/II??). This implies that something has changed regarding the permissible code alphabets. The 500Hz bandwidth puzzles me a bit - so that means that the station that is in contact with the unattended station may be using a \_different\_ bandwidth - what does this help then to help with reducing interference as the new rule was intended for ?

The only thing that I was wondering about is the 1 kHz shift issue. Without seeing the new regs, I am not sure how this applies to a multitone signal, hopefully Paul and Ian have looked at that.

--Johan

---

## 43. Re: [HFSIG:607] Re: Pitch shifting

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 21 Sep 1995 09:55:52 -0700  
To: dibene@VNET.IBM.COM  
Cc: hfsig@tapr.org  
Message-ID: <199509211655.JAA10171@servo.qualcomm.com>  
Rendered body type: text/plain

> Thanks for the explanation Phil. Your prediction seems to be already  
> true. Have you seen the new ICOM 775-DSP ? They claim SSB generation  
> by mean of a DSP-controlled PSN (Phase Shift Network?)

I'd seen the ads, but had not read them closely.

There are some other products on the market. A small company near here (Comer Communications) makes a PC plug-in card that when combined with a sound card and the right software produces a HF transceiver. The PC card is a direct conversion transceiver that produces quadrature baseband (I&Q) in a stereo jack for the sound card. The sound card can produce SSB by executing the Hilbert transform and adding or subtracting it from the untransformed data. Neat system.

Phil

---

## 44. Re:FCC regulations and multitone modems

From: tomob@ix.netcom.com (Thomas E. O'Boyle )  
Date: Thu, 21 Sep 1995 10:24:49 -0700  
To: hfsig@tapr.org  
Message-ID: <199509211724.KAA19191@ix9.ix.netcom.com>  
Rendered body type: text/plain

Well that's great news from Chris and Paul on the approvals!

I would be happy to run some tests with folks. Which EVM do I need? I'll get the units and wire up the board and the PTT circuit. I'm assuming the program will use the DTR of the serial port for the function.

I can spend about 1 to 2 hours early in the morning for testing. That might be a good time for Europe to test as well.

Let me know how I can help!

73 de Tom, N9GUN

---

#### 45. Re: [HFSIG:615] Re:FCC regulations and multitone modems

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Thu, 21 Sep 1995 11:16:03 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <46D9396609@frl.orst.edu>  
Rendered body type: text/plain

Hi Tom,

We have the code running on the Motorola EVM, called the DSP56002EVM. It costs \$150 and is available from most Motorola dealers.

Get the software and interface schematic from:

<ftp.tapr.org/tapr/SIG/HFSIG/upload/evm56k1c.zip>

Let me know of your progress and we can set up some skeds.

--Johan

---

#### 46. Re: [HFSIG:613] Re: FCC regulations and multitone modems

From: "Jon Bloom" <jbloom@usa.nai.net>  
Date: Thu, 21 Sep 1995 14:15:42 -0400  
To: hfsig@tapr.org  
Message-ID: <199509211812.OAA15910@usa.nai.net>  
Rendered body type: text/plain

Johan said:

> Although I admit that I have not seen the very latest regs, I assume that  
> the rules/subbands for data transmission apply. The new regs apparently has  
> 500Hz bandwidth for unattended data transmission, also fully legalize  
> Clover II and Pactor I (/II??). This implies that something has changed  
> regarding the permissible code alphabets. The 500Hz bandwidth puzzles me a  
> bit - so that means that the station that is in contact with the unattended  
> station may be using a \_different\_ bandwidth - what does this help then to  
> help with reducing interference as the new rule was intended for ?  
>  
> The only thing that I was wondering about is the 1 kHz shift issue. Without  
> seeing the new regs, I am not sure how this applies to a multitone signal,  
> hopefully Paul and Ian have looked at that.

1) The 500-Hz bandwidth rule applies *\*only\** to unattended stations operating outside the new "automatic" subbands. (And communication with these unattended stations can only be initiated by attended stations.)

2) Automatic stations within that subband and attended stations, on any legal RTTY/data frequency, have *\*no\** bandwidth restrictions. There is, however, a limit for FSK shift of "1 kHz between mark and space." (And when you have more than one "mark" and "space?" Hmmm...)

The best way to answer any legality question is to first determine what emission mode the signal will be. (Like F1D, J2B, etc. Maybe this thing is D1D? I can never keep these things straight.) Once you know that, you're home free.

That said, legal 3-kHz-wide signals in the CW/RTTY part of the more crowded HF bands will cause a lot of hate and discontent. I suggest restricting testing to the less crowded bands, at least at first.

Coding isn't the issue. You can--and must, by regulation--transmit ASCII, ITA2 or CCIR-476 (AMTOR) codes over any data-transparent system, including Clover and Pactor. Don't confuse data coding with line coding/modulation techniques. They are two entirely different

things.

Bottom line is, if Paul and Chris say it's OK, you can pretty much take it to the bank.

-- Jon

----

Jon Bloom, KE3Z  
jbloom@nai.net

---

#### 47. RE: [HFSIG:608] FCC regulations and multitone modems

From: "Sadowski, P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>  
Date: Thu, 21 Sep 95 09:05:00 cdt  
To: hfsig <hfsig@tapr.org>  
Message-ID: <306174FE@mailgtwy.nellis.af.mil>  
Rendered body type: text/plain

Johann

Please tell me more about acquiring the Motorola EVM. We can take this to direct E-Mail... I would have responded that way but my E-Mail system doesn't keep the original sender of E-Mail postings on the SIG.

Thanks in Advance

Allan

AH6LS

sadowskp@usafcrqs.nellis.af.mil

-----

From: hfsig

To: hfsig

Subject: [HFSIG:608] FCC regulations and multitone modems

Date: Wednesday, September 20, 1995 10:28AM

Hi all,

I did mention that we discussed the legal status of US amateur radio operator regarding the use of multitone modems on HF, at the recent DCC.

I just heard from Chris Imlay, N3AKD in Washinton that he and Paul Rinaldo, the ARRL's liason with the FCC, checked the regulations and found no reason why we could not use our multitone modems on HF, except for unattended operations, where the restriction is now 500 Hz.

This is good news. I take it from this good authority then that we can conduct our experiments on an international basis, something that would have been a problem with an STA.

So, anyone interested in doing local trials for starters? All that is needed

is a 386/486 computer running NOS, the Motorola EVM, a rather simple PTT interface, and appropriate cable to connect your transceiver to the EVM.

73's

--Johan, KC7WW

---

#### 48. Re: [HFSIG:618] RE: FCC regulations and multitone modems

From: "Jim Lynch" <k4gvo@america.net>  
Date: Fri, 22 Sep 1995 10:40:10 +0000  
To: "Sadowski, P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>, hfsig@tapr.org  
Message-ID: <199509221447.OAA05979@america.net>  
Rendered body type: text/plain

> Date: Thu, 21 Sep 1995 13:30:08 -0500  
> Reply-to: hfsig@tapr.org  
> From: "Sadowski, P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>  
> To: hfsig@tapr.org  
> Subject: [HFSIG:618] RE: FCC regulations and multitone modems

>

> Johann

>

> Please tell me more about acquiring the Motorola EVM. We can take  
> this to direct E-Mail... I would have responded that way but my E-Mail  
> system doesn't keep the original sender of E-Mail postings on the SIG.

>

> Thanks in Advance

> Allan



> AH6LS  
> sadowskp@usafcrqs.nellis.af.mil  
> -----  
PLEASE don't take it private, 'cause there are others of us  
interested too.....

Jim.  
Jim Lynch                      Fayetteville, GA              K4GVO  
'86 GL1200I                  k4gvo@america.net          44.36.34.20

---

## 49. Motorola EVM

From: rick@itron-ca.com  
Date: Fri, 22 Sep 95 09:30:36 PDT  
To: hfsig@tapr.org  
Message-ID: <Chameleon.4.01.950922093524.rick@rick.itron-ca.com>  
Rendered body type: text/plain

Call 800-845-6686 with your mastercard/visa in hand and within a few days a 56002 EVM  
will be at your doorstep.

Rick w6nzk

-----  
Rick Booth  
'95 900SS SP  
E-mail: rick@itron-ca.com  
-----

---

## 50. Re: [HFSIG:619] RE: FCC regulations and multitone modems

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 22 Sep 1995 10:07:52 -0800  
To: hfsig@tapr.org  
Message-ID: <9509221703.AA06388@ucs.orst.edu>  
Rendered body type: text/plain

Hi Jim,

>> Date: Thu, 21 Sep 1995 13:30:08 -0500  
>> Reply-to: hfsig@tapr.org  
>> From: "Sadowski,P CPT USAF CRQS/RQEN"  
>> <sadowskp@usafcrqs.nellis.af.mil>  
>> To: hfsig@tapr.org  
>> Subject: [HFSIG:618] RE: FCC regulations and multitone modems  
>  
>>  
>> Johann  
>>  
>> Please tell me more about acquiring the Motorola EVM. We can take  
>> this to direct E-Mail... I would have responded that way but my E-Mail  
>> system doesn't keep the original sender of E-Mail postings on the SIG.  
>>  
>> Thanks in Advance  
>> Allan  
>> AH6LS  
>> sadowskp@usafcrqs.nellis.af.mil  
>> -----  
>> PLEASE don't take it private, 'cause there are others of us  
>> interested too.....  
>  
>Jim.  
>Jim Lynch                      Fayetteville, GA              K4GVO  
>'86 GL1200I                  k4gvo@america.net          44.36.34.20  
>

We will try and accomodate all that may be interested.

There obviously are those that are willing and capable to help themselves. I  
have no doubt that those that are capable will be up and running shortly, so  
we have to do something to help those that need further help to get going.

We can perhaps talk TAPR into arranging for PCB's to be made and a parts kit  
put together. This would be low cost/low parts count. What do you think of  
the prospect Greg?

--Johan

---

## 51. [HFSIG:605] HFSIG at DCC

From: "Garrett, John, GARRETJ2" <GARRETJ2@btlip15.bt.co.uk>  
Date: Thu, 21 Sep 95 09:15:00 BST  
To: HFSIG@TAPR.ORG  
Message-ID: <3062F689@smtgate.comnet.bt.co.uk>  
Rendered body type: text/plain  
  
Away until 25th September. Please contact Lynn Dunnett on 01473 644859 for assistance in my absence.

---

## 52. Re: [HFSIG:621] RE: FCC regulations and multitone modems

From: Greg Jones <gjones@tenet.edu>  
Date: Fri, 22 Sep 1995 14:31:33 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199509221931.OAA09387@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

According to Johan Forrer:

> We will try and accomodate all that may be interested.  
>  
> There obviously are those that are willing and capable to help themselves. I  
> have no doubt that those that are capable will be up and running shortly, so  
> we have to do something to help those that need further help to get going.  
>  
> We can perhaps talk TAPR into arranging for PCB's to be made and a parts kit  
> put together. This would be low cost/low parts count. What do you think of  
> the prospect Greg?  
>

Not sure if everyone understands what is being disscussed here. Basically it is an interface board to the EVM.

I have someone who approached us about doing some PCB layout...so would be a good opportunity to test him out...if we can design it so that parts can be purchased at Radio Shack or DigiKey (since the parts count is low) then we could a PCB turn only...and not have to worry about parts -- which is the most time consuming aspect.

How many people are interested ? I guess we should some how get a count. Poeple should e-mail someone (who wants to volunteer) to keep a quick count.

...or might think about going to an outside source and having something made. All depends on the cost and what people want to purchase.

Sounds like a good idea.

Greg

---

## 53. (no subject)

From: (unknown sender)  
Date: (date unavailable)  
Rendered body type: text/plain  
  
Received: from p01.t0.monrotel.com by ucs.orst.edu; (5.65v3.0/1.1.8.2/24Sep94-1201PM) id AA17457; Fri, 22 Sep 1995 17:09:29 -0700  
Message-Id: <9509230009.AA17457@ucs.orst.edu>  
X-Sender: forrerj@ucs.orst.edu  
X-Mailer: Windows Eudora Version 1.4.4  
Mime-Version: 1.0  
Content-Type: text/plain; charset="us-ascii"  
Date: Fri, 22 Sep 1995 17:13:49 -0800  
To: hfsig@tapr.org  
From: forrerj@ucs.orst.edu (Johan Forrer)  
Subject: First call for EVM interface PCB

Hi Greg,

>Not sure if everyone understands what is being discussed here. Basically  
>it is an interface board to the EVM.

I changed the subject title a bit.

>

>I have someone who approached us about doing some PCB layout...so would be a  
>good opportunity to test him out...if we can design it so that parts can be  
>purchased at Radio Shack or DigiKey (since the parts count is low) then we  
>could a PCB turn only...and not have to worry about parts -- which is the  
>most time consuming aspect.

>

Danie, ZS6AWK, being an EE, has done a rather pretty PCB for his EVM. I think he would be quite willing to contribute his work - OK Danie? People should just be aware that more involved interfaces may be developed in the near future, so things are not cast in concrete. The unit that I had at the DCC has more than what is needed for our proposed tests. Note there NO modifications required on the EVM itself.

>How many people are interested ? I guess we should some how get a count.  
>People should e-mail someone (who wants to volunteer) to keep a quick count.

This is important to know before we go on. I'll be happy to take a count.

Thanks,

--Johan, KC7WW

e-mail: forrerj@ucs.orst.edu

---

## 54. Re: [HFSIG:624] First call for EVM interface PCB

From: Greg Jones <gjones@tenet.edu>  
Date: Fri, 22 Sep 1995 21:01:28 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199509230201.VAA17467@Joyce-Perkins.tenet.edu>  
Rendered body type: text/plain

Let's take a count. Talk to Danie about getting board layout files and go from there.

Basically the NRE on a board run will be \$250 (give or take) plus \$50 or so for any photoplotting that would have to be done.

It takes between two and three weeks for a turn...once everything is ready.

Just a matter of finding out how many to run. The size of run effects the per-board cost.

Greg

According to Johan Forrer:

>

> Hi Greg,

>

>

> >Not sure if everyone understands what is being discussed here. Basically  
> >it is an interface board to the EVM.

>

> I changed the subject title a bit.

>

> >

> >I have someone who approached us about doing some PCB layout...so would be a  
> >good opportunity to test him out...if we can design it so that parts can be  
> >purchased at Radio Shack or DigiKey (since the parts count is low) then we  
> >could a PCB turn only...and not have to worry about parts -- which is the  
> >most time consuming aspect.

> >

>

> Danie, ZS6AWK, being an EE, has done a rather pretty PCB for his EVM. I  
> think he would be quite willing to contribute his work - OK Danie? People  
> should just be aware that more involved interfaces may be developed in the  
> near future, so things are not cast in concrete. The unit that I had at the  
> DCC has more than what is needed for our proposed tests. Note there NO  
> modifications required on the EVM itself.

>

>

> >How many people are interested ? I guess we should some how get a count.

> >People should e-mail someone (who wants to volunteer) to keep a quick count.

>

> This is important to know before we go on. I'll be happy to take a count.

>

> Thanks,

>

> --Johan, KC7WW

>

> e-mail: forrerj@ucs.orst.edu

>

>

---

## 55. cancel

From: "Daniel R. Emblidge" <emblidge@ns.moran.com>  
Date: Fri, 22 Sep 95 22:31:04 PDT  
To: hfsig@tapr.org  
Message-ID: <Chameleon.950922223120.emblidge@emblidge.moran.com>  
Rendered body type: text/plain

-----  
Name: Daniel R. Emblidge  
E-mail: emblidge@moran.com (Daniel R. Emblidge)  
Date: 09/22/95  
Time: 22:31:04

This message was sent by Chameleon 4.5  
-----

---

## 56. Re: [HFSIG:625] Re: First call for EVM interface PCB

From: faunt@netcom.com (Doug Faunt N6TQS +1-510-655-8604)  
Date: Fri, 22 Sep 1995 20:07:32 -0700  
To: hfsig@tapr.org  
Message-ID: <199509230307.UAA14058@netcom9.netcom.com>  
Rendered body type: text/plain

Apparently I have volunteered to take a count of people interested in this board. Send mail to me, faunt@netcom.com, if you're interested in purchasing a PC board for the interface for the EVM.

Date: Fri, 22 Sep 1995 21:04:01 -0500  
Originator: hfsig@tapr.org

Let's take a count. Talk to Danie about getting board layout files and go from there.

Basically the NRE on a board run will be \$250 (give or take) plus \$50 or so for any photoplotting that would have to be done.

It takes between two and three weeks for a turn...once everything is ready.

Just a matter of finding out how many to run. The size of run effects the per-board cost.

Greg

According to Johan Forrer:

>  
> Hi Greg,  
>  
>  
> >Not sure if everyone understands what is being discussed here. Basically  
> >it is an interface board to the EVM.  
>  
> I changed the subject title a bit.  
>  
> >  
> >I have someone who approached us about doing some PCB layout...so would be a  
> >good opportunity to test him out...if we can design it so that parts can be  
> >purchased at Radio Shack or DigiKey (since the parts count is low) then we  
> >could a PCB turn only...and not have to worry about parts -- which is the  
> >most time consuming aspect.  
> >  
>  
> Danie, ZS6AWK, being an EE, has done a rather pretty PCB for his EVM. I  
> think he would be quite willing to contribute his work - OK Danie? People  
> should just be aware that more involved interfaces may be developed in the  
> near future, so things are not cast in concrete. The unit that I had at the  
> DCC has more than what is needed for our proposed tests. Note there NO  
> modifications required on the EVM itself.  
>  
>  
> >How many people are interested ? I guess we should some how get a count.  
> >People should e-mail someone (who wants to volunteer) to keep a quick count.  
>  
> This is important to know before we go on. I'll be happy to take a count.  
>  
> Thanks,  
>  
> --Johan, KC7WW

>  
> e-mail: forrerj@ucs.orst.edu  
>  
>

---

## 57. RE: [HFSIG:625] Re: First call for EVM interface PCB

From: "Sadowski, P CPT USAF CRQS/RQEN" <sadowskp@usafcrqs.nellis.af.mil>  
Date: Mon, 25 Sep 95 11:45:00 cdt  
To: hfsig <hfsig@tapr.org>  
Message-ID: <3066DA08@mailgtwy.nellis.af.mil>  
Rendered body type: text/plain

Question about the layout.... does he use "skinny" traces out to Vcc and Vdd or does he leave most of the board metal alone and just removes metal for signal isolation.. also is it single sided or dual sided? hmmm oh... what all is on "his" PCB... I got the impression that his PCB included the EVM circuitry in addition to the interface circuits.. I'm looking forward to more info...

I am interested... particularly if it's a quality board that fits in a readily available box, the parts are from digikey or Radio shack, and it's expandable..

some folks just want everything... hahahaha

Thanks in advance  
Allan  
AH6LS  
sadowskp@usafcrqs.nellis.af.mil  
-----

From: hfsig  
To: hfsig  
Subject: [HFSIG:625] Re: First call for EVM interface PCB  
Date: Friday, September 22, 1995 9:02PM

Let's take a count. Talk to Danie about getting board layout files and go from there.

> >purchased at Radio Shack or DigiKey (since the parts count is low) then we  
> >could a PCB turn only...and not have to worry about parts -- which is the  
> >most time consuming aspect.  
> >  
> >  
> Danie, ZS6AWK, being an EE, has done a rather pretty PCB for his EVM. I  
> think he would be quite willing to contribute his work - OK Danie? People  
> should just be aware that more involved interfaces may be developed in the  
> near future, so things are not cast in concrete. The unit that I had at  
the  
> DCC has more than what is needed for our proposed tests. Note there NO  
> modifications required on the EVM itself.  
>

---

## 58. Re: [HFSIG:628] Re: First call for EVM interface PCB

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Mon, 25 Sep 1995 14:08:13 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <A9BA451340@frl.orst.edu>  
Rendered body type: text/plain

Allan,

The interface is really simple: three transistor (or FET) switches for PTT, Up and Down. Two LED's (Green=heartbeat, Red=DCD). In addition I have adjustable audio attenuators to get at the audio levels. Then I also added a DIN socket on the chassis to carry the PTT and audio in/out signals. It may be a good idea to keep the connector wired in such a way that it is interchangeable with your 1200/9600 baud VHF setup, just in case you want to use it on those circuits (we won't get VHF issues here on HFSIG). As far as you can see then, there is no need for special high speed digital considerations.

I still need to get in touch with Danie and see if we can make it the same size as the EVM, with the headers aligned so that the interface plugs into the EVM to form a mother/daughterboard pair with a footprint the size of the EVM. I do not think there is much to the layout, probably a single-sided board that you can even make using the "photocopy and iron" process to make a direct etch board, that is, if you want to try it yourself.

--Johan

---

## 59. Re: [HFSIG:621] RE: FCC regulations and multitone modems

From: "Jim Lynch" <k4gvo@america.net>  
Date: Tue, 26 Sep 1995 05:19:31 +0000  
To: hfsig@tapr.org  
Message-ID: <199509260927.JAA01056@america.net>  
Rendered body type: text/plain

> Date: Fri, 22 Sep 1995 12:13:06 -0500  
> Reply-to: hfsig@tapr.org  
> From: forrerj@ucs.orst.edu (Johan Forrer)  
> To: hfsig@tapr.org  
> Subject: [HFSIG:621] RE: FCC regulations and multitone modems

> Hi Jim,  
>  
Hi, Johan..  
> >PLEASE don't take it private, 'cause there are others of us  
> >interested too.....

>  
> We will try and accomodate all that may be interested.  
>  
> There obviously are those that are willing and capable to help themselves. I  
> have no doubt that those that are capable will be up and running shortly, so  
> we have to do something to help those that need further help to get going.  
>  
> We can perhaps talk TAPR into arranging for PCB's to be made and a parts kit  
> put together. This would be low cost/low parts count. What do you think of  
> the prospect Greg?  
>  
> --Johan  
>  
Johan,

Help me here. Doesn't the EVM come on a board? How much extra  
circuitry is needed to make it play? Guess I'll have to order one  
and see..8^) Were you suggesing that this PCB is a duplicate of the  
EVM (if it has one) or an enhanced one?

Thanks,  
Jim.

>  
Jim Lynch                      Fayetteville, GA              K4GVO  
'86 GL1200I                      k4gvo@america.net              44.36.34.20

---

## 60. Making the EVM56K play

From: jalocha@vxcern.cern.ch (Pawel Jalocha)  
Date: Tue, 26 Sep 1995 14:12:49 +0100  
Message-ID: <9509261312.AA22171@dxmint.cern.ch>  
Rendered body type: text/plain

Just to explain certain facts about the EVM56K.

In principle, you DO NOT NEED ANY EXTRA CIRCUITY to make the EVM56K  
"play". When you get the bare board from Motorola, after you get familiar  
with their interface software you proceed this way:

- set the EVM56K's to 16K RAM configuration (one jumper to be moved).
- start the EVM56K.EXE and type:  
load bios.cld              (load Johan's BIOS which makes EVM56K look like DSPCARD4)  
load fft-cut.cld            (load Pawel's noise filter - or any other application)  
go 0                        (start the BIOS which then starts the application)

Your EVM56K is playing already ! Pure software you see :-)  
Put headphones in the HEAD sockets and some noisy audio into MIC  
socket and you are in the tricky business of anti-noise filters.

Now, if you with to work packet, you need to solder (\*) the second  
RS232 socket onto the EVM56K and then yuo need to move two jumpers  
and add another two. After you have done that, you can run the 1200 bps  
modem or the multi-tone modem and communicate with it in the KISS protocol.  
You stick the TRX audio into MIC socket, the OUT socket you put into  
MIC of the TRX (put an attenuator here) and here we go !  
Well, you are still missing the PTT signal but after all why not  
to use VOX ? :-)

If you want the PTT line you need to take one of the lines available on the numerous EVM56K's sockets, add a transistor, a resistor maybe and a protective Zener diode. You have a proper PTT now for the TRX.

And if you want a full blown radio interface with some other usefull stuff then think about the Johan/Danie's board. But you DO NOT NEED ANY EXTRA CIRCUITY to make the EVM56K play !

Pawel

(\*) While your soldering iron is still hot, put on the extra 28-pin IC socket for the (E)EPROM.

---

## 61. Re: [HFSIG:630] RE: FCC regulations and multitone modems

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Tue, 26 Sep 1995 08:45:19 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <BC59646B5F@frl.orst.edu>  
Rendered body type: text/plain

Hi Jim,

Let me briefly repeat what is in the QEX November 1995 article on the EVM: The EVM, or its real name: DSP56002EVM, is made by Motorola DSP division. It is intended as a low cost evaluation module and is a complete DSP system on small PCB, some 2.4 x 4 inches or so. The PCB contains a 40 MHz (20 MIPS) 56002 DSP, 32K x 24 SRAM, and a Crystal Semi CS4215 CODEC. In addition, the unit has an optional socket for a FLASH Prom. The unit hooks up to a PC's serial port at 19.2K. It is possible to load your own DSP code via the debugger, however, once debugged, one can (using supplied code) transfer the code to the FLASH Eprom and operate the unit as a free-standing application. This is how we intend operating the modem.

Now, to be able to use the modem, you will need some additional hardware to convert some TTL levels, i.e., pull in a PTT relay. In addition, you also need to scale audio levels to suit your particular tansceiver. This is the only and all the hardware that we are talking about. To enable such interfaces to be built, the EVM provides for a couple of headers (rows of pins) that allows access to the DSP's I/O pins. The interface board is thus made such that one can plug it into these headers, just like a disconnect header on a TNC.

Hopefully it is clear now that the interface board is a very sparsely populated board, with only a few transistors, diodes, capacitors, LED's and trimpots. The EVM itself is complete already with all the digital and analog circuitry to run the modem application.

Hope this helps.

--Johan

---

## 62. Re: [HFSIG:632] RE: FCC regulations and multitone modems

From: Greg Jones <gjones@tenet.edu>  
Date: Tue, 26 Sep 1995 16:41:30 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199509262141.QAA28141@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

Good overview Johan.

The board we are tlkaing about doing would be a one sided, maybe two sided board that woudl just make it easier and more stable (mechanical wise) to mount these parts, instead of bread-boarding the parts.

Cheers - Greg

According to Johan Forrer:

>  
> Hi Jim,  
>  
> Let me briefly repeat what is in the QEX November 1995 article on the EVM:  
> The EVM, or its real name: DSP56002EVM, is made by Motorola DSP division.  
> It is intended as a low cost evaluation module and is a complete DSP system  
> on small PCB, some 2.4 x 4 inches or so. The PCB contains a 40 MHz (20  
> MIPS) 56002 DSP, 32K x 24 SRAM, and a Crystal Semi CS4215 CODEC. In  
> addition, the unit has an optional socket for a FLASH Prom. The unit hooks  
> up to a PC's serial port at 19.2K. It is possible to load your own DSP

> code via the debugger, however, once debugged, one can (using supplied  
> code) transfer the code to the FLASH Eprom and operate the unit as a  
> free-standing application. This is how we intend operating the modem.  
>  
> Now, to be able to use the modem, you will need some additional hardware to  
> convert some TTL levels, i.e., pull in a PTT relay. In addition, you also  
> need to scale audio levels to suit your particular transceiver. This is  
> the only and all the hardware that we are talking about. To enable such  
> interfaces to be built, the EVM provides for a couple of headers (rows  
> of pins) that allows access to the DSP's I/O pins. The interface board is  
> thus made such that one can plug it into these headers, just like a  
> disconnect header on a TNC.  
>  
> Hopefully it is clear now that the interface board is a very sparsely  
> populated board, with only a few transistors, diodes, capacitors, LED's and  
> trimpots. The EVM itself is complete already with all the digital  
> and analog circuitry to run the modem application.  
>  
> Hope this helps.  
>  
> --Johan  
>

---

### 63. Re: [HFSIG:633] RE: FCC regulations and multitone modems

From: "Jim Lynch" <k4gvo@america.net>  
Date: Wed, 27 Sep 1995 05:42:09 +0000  
To: hfsig@tapr.org  
Message-ID: <199509270939.JAA05062@america.net>  
Rendered body type: text/plain

> > Hi Jim,  
> >  
> (snip)  
> > Hope this helps.  
> >  
> > --Johan

It did, thank you very much. I'll be ordering one today...

Jim.  
Jim Lynch                      Fayetteville, GA              K4GVO  
'86 GL1200I                  k4gvo@america.net          44.36.34.20

---

### 64. Re: [HFSIG:614] Re: Pitch shifting

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Thu, 28 Sep 1995 06:09:50 +0200  
To: hfsig@tapr.org  
Message-ID: <199509280409.GAA28629@foxbat.pix.za>  
Rendered body type: text/plain

>There are some other products on the market. A small company near here  
>(Comer Communications) makes a PC plug-in card that when combined with  
>a sound card and the right software produces a HF transceiver. The  
>PC card is a direct conversion transceiver that produces quadrature  
>baseband (I&Q) in a stereo jack for the sound card. The sound card can  
>produce SSB by executing the Hilbert transform and adding or subtracting  
>it from the untransformed data. Neat system.  
>  
>Phil  
>  
>

Phil do you perhaps have a fax or email number for the company. I am  
interested and want to get more info.

Danie ZS6AWK

---

### 65. Re: [HFSIG:635] Re: Pitch shifting

From: gwyn@paccomm.com (Gwyn Reedy, W1BEL)  
Date: Thu, 28 Sep 95 09:28:19 UTC  
To: hfsig@tapr.org  
Message-ID: <6691@paccomm.com>  
Rendered body type: text/plain

Comer Communications, Inc.  
Brian D. Comer, President  
609 Washingtonia Drive  
San Marcos, CA 92069



(619) 744-7266  
fax (619) 744-4745  
Radware BBS (619) 744-4032

-----  
Gwyn Reedy, WIBEL      gwyn@paccomm.com      WIBEL@amsat.org

+ (813) 874-2980      Fax: + (813) 872-8696      BBS: + (813) 874-3078 (V.34)

PacComm Packet Radio Systems, Inc, 4413 N. Hesperides St, Tampa, FL 33614-7618

---

## 66. EVM interface PCB layout

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Thu, 28 Sep 1995 09:00:33 -0800  
To: HFSIG@tapr.org  
Message-ID: <9509281555.AA22313@ucs.orst.edu>  
Rendered body type: text/plain

Hi,

I am in the process of finishing up the PCB layout and will post it on the HFSIG upload area. Look for it by the end of the weekend (EVMPCB0.ZIP). This will be revision 0 (Rev.0), so I would appreciate feedback so we can finalize the layout.

--Johan, KC7WW

---

## 67. Comments on Radware HF XCVR card

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 28 Sep 1995 15:17:20 -0700  
To: hfsig@tapr.org  
Message-ID: <199509282217.PAA20766@servo.qualcomm.com>  
Rendered body type: text/plain

I bought one of these last spring, but it sat on my shelf until recently when I made some time to play with it.

The card does work, but I am not overly impressed with its RX performance. It uses a direct conversion design, and as I feared it suffers from several of its inherent drawbacks.

First of all, I should say that there's \*one\* traditional DC drawback it does not suffer from: it can receive just one sideband. That's because it produces a complex (I&Q) output, and a Hilbert transformer running in the DSP sound card completes a "phasing type" receiver. This part seems to work well. The DSP filters are clearly better than anything you can get out of a passive crystal filter.

But the board does suffer from another classic drawback of direct conversion: noise. Direct conversion receivers generally have lots of baseband gain, which makes them highly susceptible to audio frequency noise. The Radware card is encased in a hefty metal shield, but it's not perfect. It still gets lots of noise from other parts of my computer. (I've not noticed any microphonics, however, probably because the VCO is a digitally controlled DDS/PLL/divider chain, not a conventional VFO with airgap tuning capacitors that make good capacitor microphones).

A dual fan card in my system induced a loud whine in the audio output. Although removing this card dropped the noise level considerably, it still picked up lesser amounts of noise from elsewhere in the system: the power supply fan, hard disk (you can hear the heads moving!) and the CPU itself (you hear the activity from the 18.2 Hz clock tick).

And if the audio gain is too high, you get feedback. And near the feedback point you get a reverberation-like effect. The echo delays are large enough to make me suspect coupling from the audio amplifier on the sound card through the power supply and into the RX board. (The delay is relevant because it indicates that the DSP filters are in the path). The feedback was definitely not acoustic -- it still occurred at the same gain level with headphones.

I put a Mini Circuits Labs broadband preamp (20-30dB gain) in front of the receiver. The extra gain pretty much solved the noise problem (the system now became external noise limited) but at the expense of significantly increased intermod. The specified IP3 of the receiver is only +16dBm, and this isn't enough with 20-30dB of extra RF gain to really resist strong shortwave broadcast stations. You could hear them

leaking through the baseband path whenever you were tuned within about 10-15 KHz on either side (ie., within the anti-alias filters before the A/D converters.)

It's a shame, because I really think this is the future of radio design and I would really like to see this card (and the company that designed it) succeed. But it just can't compete with my IC-751A, at least not yet.

Phil

---

## 68. PCB for EVM-based HF modem

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 29 Sep 1995 17:53:09 -0800  
To: hfsig@tapr.org  
Message-ID: <9509300048.AA17968@ucs.orst.edu>  
Rendered body type: text/plain

Hi,

I uploaded revision 0 of the PCB for the EVM to  
[ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb.zip](ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb.zip)

The package contains a couple of HPGL plot files for the schematic and PCB layout. Please look over the material and let me have your comments as soon as possible.

For those that are familiar with locating parts: we need three headers to interconnect the boards. They have to be about 1/2 inch high for use with square posts. I have some in my junk box but don't know where to locate these at short notice. Could someone please help with information on these items.

I followed the format as used for the DSP-93 for the transceiver to EVM connector. This is a 8-pin mini DIN.

Hope this gets you started.

73's

--Johan, KC7WW

---

## 69. Re: [HFSIG:638] Comments on Radware HF XCVR card

From: Marijan Miletic <s56a@ljutcp.hamradio.si>  
Date: Sat, 30 Sep 95 06:07:02 UTC  
To: hfsig@tapr.org  
Message-ID: <86827@ljutcp.hamradio.si>  
Rendered body type: text/plain

It is so nice to read a qualified report on Radware XCVR with DC RX performances as expected! However, +16 dBm IMD is much better than 2.700\$ new Kenwood DSP box with TS-870 label. Only 2/12 dBm was measured at 12.5 kHz spacing. There was PC plug-in RX for 0.5-1300 MHz on display at DL hamfest made by Rosseta labs. from VK. It had no antenna attached but fancy Windows interface was advertised. Good old IC-751A is not easy to beat thanks to his front end PIN AGC attenuator and decent low noise VCO...

73 de Mario, S56A, N1YU.

P.S. It is nice to see some "analog" discussion here for a change :-)

---

## 70. EVM56K's supply

From: jalocha@vxcern.cern.ch (Pawel Jalocha)  
Date: Sat, 30 Sep 1995 20:17:46 +0100  
Message-ID: <9509301917.AA25933@dxmint.cern.ch>  
Rendered body type: text/plain

The EVM56K has a little imperfection around its supply. The circuitry (DSP+RAM+CODEC, etc.) needs a single 5V supply. To make user's life easy there is a Gratz diode bridge and a voltage regulator on the board. The user should provide some 9-15V DC supply which will be then regulated down to 5V.

My experience: I have taken one cheap "universal" wall-charger type power supply. This is basically a transformer, a diode bridge with a big electrolytic capacitor. I preset it to 9V, put this into the EVM56K and the whole setup seemed to work... until I tried some of my own programs. There were some strange sounds coming out of the EVM56K and the programs would crash sometimes. I checked that there is 9V at the input to the EVM56K but the line pulsing

must have been strong enough and I think the voltage was momentarily falling below the regulator's threshold.  
I preset the power supply to 12V then, and the problem was totally gone.  
But, I noticed another one: the regulator was getting a bit too hot.  
No surprise after all, if you go from 12V down to 5V at 500-700 mA you lose some watts on the regulator.

I am saying this here to (1) warn the future EVM56K users and (2) to suggest to the designers of the extra PCB that it would be worth to put there a good power supply, maybe like in the DSPCARD4 ?  
Radio-amateurs mostly use 12-15V DC and the EVM56K gets too hot when running on this...

Pawel

---

# October 1995

Original source file: hfsig.9510

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-10/TAPR-HF\\_1995-10.txt](https://archive.org/download/TAPR-HF_1995-10/TAPR-HF_1995-10.txt)

## 1. Re: [HFSIG:641] EVM56K's supply

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sat, 30 Sep 1995 21:43:49 -0800  
To: hfsig@tapr.org  
Message-ID: <9510010438.AA22102@ucs.orst.edu>  
Rendered body type: text/plain

Dear Pawel,

>The EVM56K has a little imperfection around its supply.  
>The circuitry (DSP+RAM+CODEC, etc.) needs a single 5V supply.  
>To make user's life easy there is a Gratz diode bridge and a voltage regulator  
>on the board. The user should provide some 9-15V DC supply  
>which will be then regulated down to 5V.  
>  
>My experience: I have taken one cheap "universal" wall-charger type power  
>supply. This is basically a transformer, a diode bridge with a big electrolytic  
>capacitor. I preset it to 9V, put this into the EVM56K and the whole setup  
>seemed to work... until I tried some of my own programs. There were some  
>strange sounds coming out of the EVM56K and the programs would crash sometimes.  
>I checked that there is 9V at the input to the EVM56K but the line pulsing  
>must have been strong enough and I think the voltage was momentary falling  
>below the regulator's threshold.  
>I preset the power supply to 12V then, and the problem was totally gone.  
>But, I noticed another one: the regulator was getting a bit too hot.  
>No surprise after all, if you go from 12V down to 5V at 500-700 mA you loose  
>some watts on the regulator.  
>  
>I am saying this here to (1) warn the future EVM56K users and (2) to suggest  
>to the designers of the extra PCB that it would be worth to put there  
>a good power supply, maybe like in the DSPCARD4 ?  
>Radio-amateurs mostly use 12-15V DC and the EVM56K gets too hot when running  
>on this...  
>  
>Pawel  
>

Yes, that power input circuit on the EVM was a bad idea. I did tell the folks at Motorola that and they acknowledged. The best idea in my opinion, is to remove the diode bridge, and feed 8V DC directly to the regulator. In addition pay attention to where the supply ground is attached.

Unfortunately, many of the guys that will be participating in testing the HF modem, is not technical types. They need a simple interface that just plugs into the EVM and they do not want to do any modifications to the EVM at all. I have also just heard from Danie that the newer EVM's now has a ceramic resonator instead of a crystal. It also now have the second RS-232 socket soldered in. So what else did they change?

Have you had a chance yet to look at the schematic and PCB layout that I uploaded to [ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb0.zip](ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb0.zip) ?? I would appreciate some feedback so we can finalize it.

Hope you are enjoying your stay at CERN?

--Johan

---

## 2. Re: [HFSIG:638] Comments on Radware HF XCVR card

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sun, 1 Oct 1995 09:11:07 +0200  
To: hfsig@tapr.org  
Message-ID: <199510010711.JAA11587@concorde.pix.za>  
Rendered body type: text/plain

Phil interesting about the DC receiver card for the PC. I suggest they made the following mistakes which could be rectified of course:

1) The receiver card should be a standalone unit ie not INSIDE the noisy PC with its OWN PSU.

2) The architecture should be at least a dual conversion with more gain in the RF and first IF stages as you mentioned. Die LO's MUST be DDS based ie

Q2334 hi..

3) There could be a KISS interface to the receiver (now standalone) to change its parameters ie frequency, mode etc

4) It should be connected to a DSP56002EVM ! with its I&Q outputs going into Line R & L on the EVM. In fact the KISS interface could be to the EVM and the EVM could control the RF side of the receiver card. One could now also implement digital communications experiments and not only listen to boring voice conversations.

What do you guys say ? Sounds like a project for hfsig...?

73 Danie ZS6AWK

---

### 3. Re: [HFSIG:641] EVM56K's supply

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sun, 1 Oct 1995 11:15:13 +0200  
To: hfsig@tapr.org  
Message-ID: <199510010915.LAA13000@concorde.pix.za>  
Rendered body type: text/plain

>I am saying this here to (1) warn the future EVM56K users and (2) to suggest  
>to the designers of the extra PCB that it would be worth to put there  
>a good power supply, maybe like in the DSPCARD4 ?  
>Radio-amateurs mostly use 12-15V DC and the EVM56K gets too hot when running  
>on this...  
>  
>Pawel  
>  
>

Thanks Pawel for the input. I think one must use a switchmode as they use on the CARD4. Or may be one of these small switch modes from Powertran; they give a nice cool +5V from I think about 8 to 28V. Not sure about the input spec. Running a LM7805 from a high voltage is NOT a good idea.

I agree on the adaptor stuff, I measure a rather large rippel voltage on the supply when my interface is plugged onto the EVM

73 Danie ZS6AWK

---

### 4. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 13:00:09  
To: hfsig@tapr.org  
Message-ID: <199510021803.NAA31686@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

### 5. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 13:05:05  
To: hfsig@tapr.org  
Message-ID: <199510021805.NAA31905@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

### 6. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 13:55:04  
To: hfsig@tapr.org  
Message-ID: <199510021856.NAA00717@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 7. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 13:55:05  
To: hfsig@tapr.org  
Message-ID: <199510021856.NAA00732@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 8. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 14:30:04  
To: hfsig@tapr.org  
Message-ID: <199510021930.OAA01440@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 9. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 14:30:08  
To: hfsig@tapr.org  
Message-ID: <199510021930.OAA01453@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 10. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 14:55:03  
To: hfsig@tapr.org  
Message-ID: <199510021959.OAA02238@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 11. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 15:00:03  
To: hfsig@tapr.org  
Message-ID: <199510022010.PAA02558@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 12. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 15:30:05  
To: hfsig@tapr.org  
Message-ID: <199510022030.PAA02991@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 13. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 15:35:03  
To: hfsig@tapr.org  
Message-ID: <199510022038.PAA03342@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 14. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 16:00:07  
To: hfsig@tapr.org  
Message-ID: <199510022101.QAA03924@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 15. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 16:05:09  
To: hfsig@tapr.org  
Message-ID: <199510022107.QAA04193@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 16. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 16:40:05  
To: hfsig@tapr.org  
Message-ID: <199510022140.QAA05203@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 17. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 16:40:07  
To: hfsig@tapr.org  
Message-ID: <199510022140.QAA05216@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 18. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 17:25:06  
To: hfsig@tapr.org  
Message-ID: <199510022225.RAA06272@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 19. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 17:25:08  
To: hfsig@tapr.org  
Message-ID: <199510022225.RAA06276@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 20. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 18:05:04  
To: hfsig@tapr.org  
Message-ID: <199510022305.SAA07327@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 21. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 18:25:04  
To: hfsig@tapr.org  
Message-ID: <199510022325.SAA07653@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 22. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 18:25:05  
To: hfsig@tapr.org  
Message-ID: <199510022326.SAA07665@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.



## 23. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 18:45:03  
To: hfsig@tapr.org  
Message-ID: <199510022345.SAA08348@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 24. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 19:00:05  
To: hfsig@tapr.org  
Message-ID: <199510030001.TAA08809@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 25. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 19:00:07  
To: hfsig@tapr.org  
Message-ID: <199510030001.TAA08817@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 26. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 19:25:04  
To: hfsig@tapr.org  
Message-ID: <199510030027.TAA09741@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 27. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 19:30:08  
To: hfsig@tapr.org  
Message-ID: <199510030030.TAA09854@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 28. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 19:35:15  
To: hfsig@tapr.org  
Message-ID: <199510030035.TAA10151@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 29. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 19:55:07  
To: hfsig@tapr.org  
Message-ID: <199510030056.TAA10661@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 30. Unrecognized recipient in message

From: tsxbbs.smtp@nashville.com  
Date: Mon, 2 Oct 95 20:00:25  
To: hfsig@tapr.org  
Message-ID: <199510030103.UAA10735@sys1.tapr.org>  
Rendered body type: text/plain

You recently sent a message to e.johnson@nashville.com which could not be delivered. There is no person with this name at The Nashville Exchange BBS 615-383-0727. Typically, this is the result of a slight misspelling or the omission of a period between the first and last names. Your message was delivered to the mail clerk, Ben Cunningham, since your intended recipient was not recognized.

---

## 31. Re: [HFSIG:642] Re: EVM56K's supply

From: "Jim Lynch" <k4gvo@america.net>  
Date: Tue, 3 Oct 1995 05:26:05 +0000  
To: hfsig@tapr.org  
Message-ID: <199510030924.JAA09690@america.net>  
Rendered body type: text/plain

> Date: Mon, 2 Oct 1995 12:45:54 -0500  
> Reply-to: hfsig@tapr.org  
> From: forrerj@ucs.orst.edu (Johan Forrer)  
> To: hfsig@tapr.org  
> Subject: [HFSIG:642] Re: EVM56K's supply

> Dear Pawel,  
>  
> >The EVM56K has a little imperfection around its supply.  
> >The circuitry (DSP+RAM+CODEC, etc.) needs a single 5V supply.  
> >To make user's life easy there is a Gratz diode bridge and a voltage regulator  
> >on the board. The user should provide some 9-15V DC supply  
> >which will be then regulated down to 5V.  
> >  
> >My experience: I have taken one cheap "universal" wall-charger type power  
> >supply. This is basically a transformer, a diode bridge with a big electrolytic  
> >capacitor. I preset it to 9V, put this into the EVM56K and the whole setup  
> >seemed to work... until I tried some of my own programs. There were some  
> >strange sounds coming out of the EVM56K and the programs would crash sometimes.  
> >I checked that there is 9V at the input to the EVM56K but the line pulsing  
> >must have been strong enough and I think the voltage was momentarily falling  
> >below the regulator's threshold.  
> >I preset the power supply to 12V then, and the problem was totally gone.  
> >But, I noticed another one: the regulator was getting a bit too hot.  
> >No surprise after all, if you go from 12V down to 5V at 500-700 mA you loose  
> >some watts on the regulator.

My calculations show that a 3.9 ohm, 2 watt resistor might help that.  
What do you think?

> >  
> >I am saying this here to (1) warn the future EVM56K users and (2) to suggest  
> >to the designers of the extra PCB that it would be worth to put there  
> >a good power supply, maybe like in the DSPCARD4 ?  
> >Radio-amateurs mostly use 12-15V DC and the EVM56K gets too hot when running  
> >on this...  
> >

> >Pawel  
> >  
>  
> Yes, that power input circuit on the EVM was a bad idea. I did tell the  
> folks at Motorola that and they acknowledged. The best idea in my opinion,  
> is to remove the diode bridge, and feed 8V DC directly to the regulator. In  
> addition pay attention to where the supply ground is attached.  
>  
> Unfortunately, many of the guys that will be participating in testing the HF  
> modem, is not technical types. They need a simple interface that just plugs  
> into the EVM and they do not want to do any modifications to the EVM at all.  
> I have also just heard from Danie that the newer EVM's now has a ceramic  
> resonator instead of a crystal. It also now have the second RS-232 socket  
> soldered in. So what else did they change?  
>  
> Have you had a chance yet to look at the schematic and PCB layout that I  
> uploaded to ftp.tapir.org/tapir/SIG/hfsig/upload/evmpcb0.zip ?? I would  
> appreciate some feedback so we can finalize it.  
>  
> Hope you are enjoying your stay at CERN?  
>  
> --Johan  
>  
>  
Jim Lynch                      Fayetteville, GA              K4GVO  
'86 GL1200I                  k4gvo@america.net            44.36.34.20

---

### 32. Re: [HFSIG:640] Re: Comments on Radware HF XCVR card

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 3 Oct 1995 21:45:45 -0700  
To: hfsig@tapir.org  
Message-ID: <199510040445.VAA22104@servo.qualcomm.com>  
Rendered body type: text/plain  
  
>performances as expected! However, +16 dBm IMD is much better than  
  
That was the specified figure; I didn't measure it. It was also  
specified for 100 Khz spacing, which seems a bit wide. I noticed in  
a qualitative test that the IM got worse when the spacing was less  
than 100 Khz -- this is probably the bandwidth of the low pass filters  
following the mixers.  
  
Phil

---

### 33. Re: [HFSIG:672] Re: EVM56K's supply

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Wed, 4 Oct 1995 06:52:00 +0200  
To: hfsig@tapir.org  
Message-ID: <199510040452.GAA01511@concorde.pix.za>  
Rendered body type: text/plain  
  
Regarding the power-on-reset problem on the EVM:  
  
I must say I am still a bit skeptical about the remarks about the EVM psu.  
All the programs I wrote, (admittedly they are smaller than Pawel's programs  
combined with bios.asm), run with out a hitch. I tested some for hours on  
end. I even implemented my own 'heart beat' function. I think Johan might be  
correct about suspecting the CODEC and relevant stuff.  
  
The debate is interesting because the fault is difficult to trace...and we  
still don't have an answer. My approach has been to re-write the bios.asm  
into my own minibios.asm It is currently growing slowly as I add more  
'copied & modified' functions from bios.asm Sooner or later I must pick up  
the culprit as I integrate it into my minibios.asm  
  
73 danie zs6awk

---

### 34. Re: [HFSIG:674] Re: EVM56K's supply

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Wed, 4 Oct 1995 08:33:16 PST8PDT  
To: hfsig@tapir.org  
Message-ID: <17C2BA4442B@frl.orst.edu>  
Rendered body type: text/plain  
  
Hi,  
  
One further bit that I may add to the EVM supply issue:

I have been using an LC filter spliced into the supply lead. It consists of a 100uH choke (on the input side) and a 2200uF electrolytic (on the output side). The wall-wart is marked as being 8VDC, however it puts out 15VDC unloaded. With the LC filter there is very little ripple left - output is around 9VDC when drawing the EVM's load - the heatsink is just warm to the touch even after hours of operation. I put the covered the filter in a piece of shrink tubing, still looks ugly, but it seems to work.

There is a similar thread on Usenet "COMP.DSP" regarding the EVM's supply. One poster mentioned that later versions of the EVM has a fatter filter cap of 470uF replacing the 100uF, however that is not sufficient - he found that he had to increase that to 1000uF at which he found that even the AC supply seemed to work OK.

--Johan

---

## 35. Re: [HFSIG:643] Re: Comments on Radware HF XCVR card

From: Phil Karn <karn@qualcomm.com>  
Date: Wed, 4 Oct 1995 10:44:38 -0700  
To: hfsig@tapr.org  
Message-ID: <199510041744.KAA28967@servo.qualcomm.com>  
Rendered body type: text/plain

>1) The receiver card should be a standalone unit ie not INSIDE the noisy PC  
>with its OWN PSU.

There is a massive aluminum shield on both sides of the board. But RF coupling doesn't seem to be the problem; I tried removing the shield entirely and it seemed to make relatively little difference. A much larger increase in noise occurred when I disconnected the big DC bypass cap that was connected from the input of the 7805 regulator to ground (it was grounded to a screw holding the shield in place so I had to remove it anyway). Touching the cap to ground noticeably reduced the noise even with the shield removed.

I conclude from this that the primary noise path is conduction through the power line from the PC. So yes, moving the receiver to a separate box would help but mainly because it would then have its own power supply.

>2) The architecture should be at least a dual conversion with more gain in  
>the RF and first IF stages as you mentioned. Die LO's MUST be DDS based ie  
>Q2334 hi..

I don't know if more RF gain would be warranted as that would worsen the intermod performance even further. I've had some extensive discussions with a colleague who's an RF designer. He has frequently bent my ear about the importance of proper mixer design, and now I grok what he was talking about. If you screw up on the first mixer, there's nothing you can do elsewhere to fix it.

The LO in this unit is DDS-based, but it's followed by a PLL multiplier and divider. I have heard this is actually fairly standard practice, and is not by itself anything to complain about. The LO actually seems pretty good in this unit, at least compared to the noise and intermod problems.

>3) There could be a KISS interface to the receiver (now standalone) to  
>change its parameters ie frequency, mode etc

Well, KISS was designed for a totally different application so this is a nonsequitor, but yes, you could probably use a serial port interface of some sort. My only concern is that such ports are often only fast enough for the "usual" use of a transceiver, and might not be fast enough to support the frequency hopped spread spectrum applications I had in mind.

>4) It should be connected to a DSP56002EVM ! with its I&Q outputs going into  
>Line R & L on the EVM. In fact the KISS interface could be to the EVM and  
>the EVM could control the RF side of the receiver card. One could now also  
>implement digital communications experiments and not only listen to boring  
>voice conversations.

You could connect it to anything you like. In this sense I think the Radware people did exactly the right thing by using existing PC sound cards. They did use a DSP sound card instead of native signal processing, but at least they used something that you can buy on the mass market.

Phil

### 36. Updated EVM Interface

From: "Johan Forrer" <FORRERJ@frl.orst.edu>  
Date: Thu, 5 Oct 1995 09:46:27 PST8PDT  
To: hfsig@tapr.org  
Message-ID: <19564995681@frl.orst.edu>  
Rendered body type: text/plain

Hi,

I have made a few cosmetic changes to the EVM radio interface  
and placed it at ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb1.zip

Would appreciate further feedback.

73's

--Johan

---

### 37. Re: [HFSIG:677] Updated EVM Interface

From: John Koster <jkoster@tenet.edu>  
Date: Thu, 5 Oct 1995 20:38:02 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199510060138.UAA29497@Gayle-Gaston.tenet.edu>  
Rendered body type: text/plain

Johan, Greg has asked me to get involved from the board house interface  
aspect. Was wondering if you are able to generate Gerber files for the  
boards? I was able to pull in your original posting and view the HP  
format with some minor 'funny stuff' showing up here and there. It is  
possible that the board house can use the HP format, but I'll have to ask.

I haven't looked at your update yet, but will go get it shortly.

73,

John

w9ddd@tapr.org

According to Johan Forrer:

>  
> Hi,  
>  
> I have made a few cosmetic changes to the EVM radio interface  
> and placed it at ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb1.zip  
>  
> Would appreciate further feedback.  
>  
> 73's  
>  
> --Johan  
>

---

### 38. Re: [HFSIG:678] Re: EVM Interface

From: "Jim Lynch" <k4gvo@america.net>  
Date: Fri, 6 Oct 1995 05:28:47 +0000  
To: hfsig@tapr.org  
Message-ID: <199510060927.JAA15182@america.net>  
Rendered body type: text/plain

OK, I've got a very impressive EVM in my hot little hands now. Is  
there a description of the interface/schematic? A list of the  
functions of the pins would probably be enough. I've got a pretty  
well supplied junk box and a hot soldering iron.

Some of my 7400s are code dated 1974! I  
wonder if 20 year old IC still work? Rhetorical question....forget  
it.

I'm going to be using a FT-990 for HF.

Jim.  
Jim Lynch                      Fayetteville, GA              K4GVO  
'86 GL1200I                  k4gvo@america.net              44.36.34.20

---

### 39. Re: [HFSIG:678] Re: Updated EVM Interface

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 06 Oct 1995 11:04:35 -0800

To: hfsig@tapr.org  
Message-ID: <9510061759.AA15338@ucs.orst.edu>  
Rendered body type: text/plain

Hi John,

>Johan, Greg has asked me to get involved from the board house interface  
>aspect. Was wondering if you are able to generate Gerber files for the  
>boards? I was able to pull in your original posting and view the HP  
>format with some minor 'funny stuff' showing up here and there. It is  
>possible that the board house can use the HP format, but I'll have to ask.  
>  
>I haven't looked at your update yet, but will go get it shortly.  
>  
>73,  
>  
>John  
>  
>w9ddd@tapr.org  
>  
>According to Johan Forrer:  
>>  
>> Hi,  
>>  
>> I have made a few cosmetic changes to the EVM radio interface  
>> and placed it at ftp.tapr.org/tapr/SIG/hfsig/upload/evmpcb1.zip  
>>  
>> Would appreciate further feedback.  
>>  
>> 73's  
>>  
>> --Johan  
>>  
>  
>

Pleased to hear that you would be interested to help with the board.

The PCB layout package that I use (Wintek, SmartArtwork) unfortunately does not produce Gerber files, although they do offer such an add-on option that costs too much for a poor ham like me. In the past I have worked with board houses that did the translation to Gerber files in-house with no additional charge, and I suspect that it may just be possible for the folks that you work with may offer that service too. In any case, the board is quite simple and sparsely populated that it wouldn't take much to re-draw it with another program package such as Tango to output Gerber files.

I am pretty much contented with what is on the PCB except for the 8-pin mini DIN. If someone can check out the physical dimensions and pin placement, that would be help. Otherwise, I will be making a prototype using the laser printer iron-on method and ordering the parts to make sure that all fits as intended. You may want to hang on till we have all checked out, but it is up to you of course.

73's

--Johan

---

#### 40. Re: [HFSIG:679] Re: EVM Interface

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 06 Oct 1995 11:04:55 -0800  
To: hfsig@tapr.org  
Message-ID: <9510061759.AA19149@ucs.orst.edu>  
Rendered body type: text/plain

Hi Jim,

>OK, I've got a very impressive EVM in my hot little hands now. Is  
>there a description of the interface/schematic? A list of the  
>functions of the pins would probably be enough. I've got a pretty  
>well supplied junk box and a hot soldering iron.  
>  
> Some of my 7400s are code dated 1974! I  
>wonder if 20 year old IC still work? Rhetorical question....forget  
>it.  
>  
>I'm going to be using a FT-990 for HF.  
>  
>Jim.  
>Jim Lynch                      Fayetteville, GA                      K4GV0

>'86 GL1200I            k4gvo@america.net        44.36.34.20  
>

I don't have anything yet besides what is in the package. The schematic is pretty much self-explanatory and is a good place to start. Plot/print the files. If you look at the J7 header - that gives one access to the DSP's output port. At the moment we only need three output pins, B0 (PTT), B1 (Up) and B2 (Down). These signals are routed to a ULN chip that contains the switching transistors. There are two status LED's in addition that are on B13 and B14. You would need to bring them up to the front panel so you can see DCD and the modem's "heartbeat". Besides that, there are two audio potentiometers to adjust the audio levels for the transceiver.

Does it make sense?

--Johan

---

## 41. TORLIB

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 06 Oct 1995 14:53:05 -0800  
To: hfsig@tapr.org  
Message-ID: <9510062147.AA24843@ucs.orst.edu>  
Rendered body type: text/plain

Hi,

I posted a library of source code and makefiles for building AMTOR and PACTOR programs that will run on a PC. This is the code version that supports PSA sound card modems. Check the HFSIG upload area for TORLIB1.ZIP.

You will ONLY be interested in this if you are a very familiar with programming in C and assembly and want to write your own code to operate on AMTOR or PACTOR. If you have a PSA soundcard and only want to use the current executables, use PSATOR23.ZIP.

The software is copyleft and released under the general GNU license. I don't intend advertising this library much further, hope that others on HFSIG may find it of use.

Enjoy.

--Johan

---

## 42. Re: [HFSIG:681] Re: EVM Interface

From: "Jim Lynch" <k4gvo@america.net>  
Date: Sat, 7 Oct 1995 06:43:49 +0000  
To: forrerj@ucs.orst.edu (Johan Forrer), hfsig@tapr.org  
Message-ID: <199510071042.KAA16643@america.net>  
Rendered body type: text/plain

> Date:            Fri, 6 Oct 1995 13:18:20 -0500  
> Reply-to:        hfsig@tapr.org  
> From:            forrerj@ucs.orst.edu (Johan Forrer)  
> To:              hfsig@tapr.org  
> Subject:         [HFSIG:681] Re: EVM Interface

> Hi Jim,

Hi, Johan,

>

(valued information deleted)

> Does it make sense?

>

> --Johan

>

>

OK, I got evmpcb1.zip.

Sense, yes, but does anyone know of a way to convert hpgl to something else? I've got a PostScript printer available as well as a HP LaserJet Series II, but alas no hp plotter. Years ago I heard of a conversion program to convert from hpgl to hp laserjet, but don't remember where.

Another question, Johan, I'm not familiar with the ULN chip. Where do you suggest I look for it? I'll buy a handful and offer them to others that might not need enough stuff to make up a minimum order.

If I can get the schematic printed, I'll offer it to others for

shipping charges.

Thanks,

Jim.

Jim Lynch  
'86 GL1200I

Fayetteville, GA  
k4gvo@america.net

K4GVO  
44.36.34.20

---

### 43. Re: [HFSIG:683] Re: EVM Interface

From: "Jon Bloom" <jbloom@usa.nai.net>  
Date: Sat, 7 Oct 1995 08:15:06 -0400  
To: hfsig@tapr.org  
Message-ID: <199510071211.IAA11912@usa.nai.net>  
Rendered body type: text/plain

> Date: Sat, 7 Oct 1995 05:57:19 -0500  
> Reply-to: hfsig@tapr.org  
> From: "Jim Lynch" <k4gvo@america.net>  
> To: hfsig@tapr.org  
> Subject: [HFSIG:683] Re: EVM Interface

> Sense, yes, but does anyone know of a way to convert hpgl to  
> something else? I've got a PostScript printer available as well as a  
> HP LaserJet Series II, but alas no hp plotter. Years ago I heard of  
> a conversion program to convert from hpgl to hp laserjet, but don't  
> remember where.

Try ftp://oak.oakland.edu/SimTel/msdos/plot/prtg1152.zip

I've used this PrintGL program for several years with great success.  
It supports various printers, including both PCL and Postscript.

Jon, KE3Z

jbloom@nai.net

----

Jon Bloom, KE3Z  
jbloom@nai.net

---

### 44. Re: [HFSIG:683] Re: EVM Interface

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Sat, 7 Oct 1995 17:10:43 +0100 (MET)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.951007170909.9486A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Sat, 7 Oct 1995, Jim Lynch wrote:

> Sense, yes, but does anyone know of a way to convert hpgl to  
> something else? I've got a PostScript printer available as well as a  
> HP LaserJet Series II, but alas no hp plotter. Years ago I heard of  
> a conversion program to convert from hpgl to hp laserjet, but don't  
> remember where.  
>

There is a program that can do this, it is called printgl.

73's Frode

|                                |        |                        |
|--------------------------------|--------|------------------------|
| Frode Weierud                  | Phone  | : 41 22 7674794        |
| CERN, SL                       | Fax    | : 41 22 7679185        |
| CH-1211 Geneva 23, Switzerland | E-mail | : frode@dxcern.cern.ch |

---

### 45. Re: [HFSIG:683] Re: EVM Interface

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sat, 07 Oct 1995 19:28:13 -0800  
To: hfsig@tapr.org  
Message-ID: <9510080228.AA15063@ucs.orst.edu>  
Rendered body type: text/plain

Hi Jim,

>> Date: Fri, 6 Oct 1995 13:18:20 -0500  
>> Reply-to: hfsig@tapr.org  
>> From: forrerj@ucs.orst.edu (Johan Forrer)  
>> To: hfsig@tapr.org  
>> Subject: [HFSIG:681] Re: EVM Interface  
>



```

>> Hi Jim,
>
>Hi, Johan,
>>
>(valued information deleted)
>> Does it make sense?
>>
>> --Johan
>>
>>
>OK, I got evmpcb1.zip.
>
>Sense, yes, but does anyone know of a way to convert hpgl to
>something else? I've got a PostScript printer available as well as a
>HP LaserJet Series II, but alas no hp plotter. Years ago I heard of
>a conversion program to convert from hpgl to hp laserjet, but don't
>remember where.
>

```

I have been using the PRINTGL program with good success too.

```

>Another question, Johan, I'm not familiar with the ULN chip. Where
>do you suggest I look for it? I'll buy a handful and offer them to
>others that might not need enough stuff to make up a minimum order.

```

I looked through some catalogs and found some of the parts - if anyone else knows of other sources, or better prices, please along details.

Mouser Electronics (800-346-6873)

```

ULN2803A - SGS-Thompson $1.06 (1-10)
161-2308 - 8-pin Mini DIN Socket $1.96
171-2508 - 8-pin Mini DIN Plug $1.43

```

Newark Electronics (Look for dealer in your area)

```

95F6680 - (Berg 68683-308 2x8 positions) $4.29 (1-49)
50F8766 - (Berg 68683-315 2x15 positions) $6.97 (1-49)

```

Just a note about the inter-board connectors:

```

These have to be mounted on the solder-side of the PCB in order to plug into
the EVM, i.e., the interface board is mounted on top of the EVM. If you
fabricate your own single-sided PCB, you would need some way of soldering on
the component side of the PCB where there are, of course, no solder pads. An
old trick is to thread short pieces of bare wire-wrap wire through the
connector's holes to the component side, solder- tack them to the solder
pads on the solder side, and then push the connector into place. Then wrap
each solder tail's wire and complete the soldering. A bit of work, but it
looks quite neat and does the trick. If you have a through-hole plated
board, of course, the through-hole connection have already been made for you.

```

Before you solder though, make sure that there is sufficient clearance between the top board and the EVM when they go together - the EVM's heat sink and the screw-connector may be in the way. From the dimensions of the Berg connectors in the catalog, this may well be the case. One either have to make a small cut-out in the interface board for these components to clear, or, if someone can point out use to a taller version of such a connector - those does exists as I have some in my junk box.

Can anyone suggest a taller connector?

```

>
>If I can get the schematic printed, I'll offer it to others for
>shipping charges.
>
>Thanks,
>Jim.
>Jim Lynch Fayetteville, GA K4GVO
>'86 GL1200I k4gvo@america.net 44.36.34.20
>

```

Thanks for the offer Jim.  
--Johan

## 46. Re: [HFSIG:686] Re: EVM Interface

From: "Jim Lynch" <k4gvo@america.net>  
Date: Mon, 9 Oct 1995 05:31:53 +0000  
To: hfsig@tapr.org  
Message-ID: <199510090931.JAA03508@america.net>  
Rendered body type: text/plain

OFFER OPEN TO USA residents only at this time. I've got to check to see what postage to the rest of NA might be.

I've printed off Johan's schematic and board drawings for the EVM interface. If you would like a copy please send a business-sized SASE to:

Jim Lynch K4GVO  
128 Carrollwood Drive  
Fayetteville, GA 30214

The board layouts did not print 1:1. They appear to be about 30% larger than life. I guess the program enlarged them to fit on a 8.5 x 11 page.

There are 4 pages. If you don't want them sent folded, send me a large envelope. Remember the postage is more for envelopes larger than letter size.

If you don't want to bother with the SASE, send me \$1 and I'll put them in a large envelope for you.

73,  
Jim.  
Jim Lynch                      Fayetteville, GA                      K4GVO  
'86 GL1200I                      k4gvo@america.net                      44.36.34.20

---

#### 47. Re: [HFSIG:684] Re: EVM Interface

From: Alan Bloom <alanb@tsnake2.sr.hp.com>  
Date: Mon, 9 Oct 1995 15:18:36 -0800 (PDT)  
To: hfsig@tapr.org (Jon Bloom)  
Message-ID: <199510092218.AA081877117@tsnake2.sr.hp.com>  
Rendered body type: text/plain

>  
> > Sense, yes, but does anyone know of a way to convert hpgl to  
> > something else? I've got a PostScript printer available as well as a  
> > HP LaserJet Series II, but alas no hp plotter. Years ago I heard of  
> > a conversion program to convert from hpgl to hp laserjet, but don't  
> > remember where.  
>  
> Try ftp://oak.oakland.edu/SimTel/msdos/plot/prtgl152.zip  
>  
> I've used this PrintGL program for several years with great success.  
> It supports various printers, including both PCL and Postscript.

Also, later-model HP laser printers (Laserjet III and beyond) support HPGL internally.

AL N1AL

---

#### 48. Re: [HFSIG:686] Re: EVM Interface

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Tue, 10 Oct 1995 13:22:35 +0100 (MET)  
To: hfsig@tapr.org  
Message-ID: <Pine.ULT.3.91.951010132035.3123A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

On Sat, 7 Oct 1995, Johan Forrer wrote:

> Before you solder though, make sure that there is sufficient clearance  
> between the top board and the EVM when they go together - the EVM's heat  
> sink and the screw-connector may be in the way. From the dimensions of the  
> Berg connectors in the catalog, this may well be the case. One either have  
> to make a small cut-out in the interface board for these components to  
> clear, or, if someone can point out use to a taller version of such a  
> connector - those does exists as I have some in my junk box.  
>  
> Can anyone suggest a taller connector?

Does this type of connector exist in a wire-wrap version with long pins?  
If so this might be the solution to the spacing problem.

Frode

Frode Weierud                      Phone : 41 22 7674794  
CERN, SL                              Fax : 41 22 7679185  
CH-1211 Geneva 23, Switzerland   E-mail : frode@dxcern.cern.ch

---

#### 49. ARLB094 Digital rules clarified (fwd)

From: Greg Jones <gjones@tenet.edu>  
Date: Thu, 12 Oct 1995 16:52:25 -0500 (CDT)  
To: hfsig@tapr.org (HF SIG mailing)  
Message-ID: <199510122152.QAA09165@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

Good news HF Digital People. This would mean that publishing a communications method in the PSR or better yet the ARRL and TAPR Digital Communications Conference (where the below mentioned methods were published in 1994) makes it a blessed thing.

Cheers - Greg

According to wlaw@arrl.org:  
>From daemon Thu Oct 12 16:22:17 1995  
Subject: ARLB094 Digital rules clarified  
To: QST@sys1.tapr.org  
From: wlaw@arrl.org  
Organization: American Radio Relay League  
Date: Thu, 12 Oct 1995 17:21:31 EDT  
Message-ID: <\$arlb094.1995@ampr.org>

SB QST @ ARL \$ARLB094  
ARLB094 Digital rules clarified

ZCZC AG61  
QST de W1AW  
ARRL Bulletin 94 ARLB094  
>From ARRL Headquarters  
Newington CT October 12, 1995  
To all radio amateurs

SB QST ARL ARLB094  
ARLB094 Digital rules clarified

The FCC, responding to a request from the ARRL, has clarified its rules regarding Amateur Radio stations' use of digital codes.

The ARRL, in a letter to the Commission, said that some amateurs had expressed concern about the propriety of using the CLOVER, G-TOR, and PacTOR codes on the HF bands (below 30 MHz), because FCC rule 97.309(a) 'appears to authorize' only Baudot, AMTOR, and ASCII codes. The League told the Commission that it had worked with the developers of CLOVER, G-TOR, and PacTOR to document the technical characteristics of those codes, and asked the FCC to remove any doubt about their legality on the HF bands.

The FCC agreed, saying that the use of the three modes does not conflict with its objective of preventing the use of codes or ciphers intended to obscure the meaning of the communication.

As a result, the FCC added a paragraph (4) to Part 97.309(a) to read:

(4) An amateur station transmitting a RTTY or data emission using a digital code specified in this paragraph may use any technique whose technical characteristics have been documented publicly, such as CLOVER, G-TOR, or PacTOR, for the purpose of facilitating communications.

NNNN  
/EX

---

#### 50. [FCCREG:339] FCC dd101195.txt 10/11 Daily Digest (fwd)

From: Greg Jones <gjones@tenet.edu>  
Date: Fri, 13 Oct 1995 15:04:52 -0500 (CDT)  
To: hfsig@tapr.org (HF SIG mailing)  
Message-ID: <199510132004.PAA28326@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain  
Subject: [FCCREG:339] FCC dd101195.txt 10/11 Daily Digest

...

AMATEUR SERVICE RULES. Clarified that amateur stations may use any digital code that has its technical characteristics publicly documented. (By Order [DA 95-2106] adopted October 2 by the Chief, Wireless Telecommunications Bureau)

...

---

## 51. Re: [HFSIG:690] ARLB094 Digital rules clarified (fwd)

From: Phil Karn <karn@qualcomm.com>  
Date: Mon, 16 Oct 1995 18:28:10 -0700  
To: hfsig@tapr.org  
Message-ID: <199510170128.SAA04530@servo.qualcomm.com>  
Rendered body type: text/plain

All right!!! Finally we can stop having to defend our stuff against all those complaints of illegality...

Phil

---

## 52. Re: [HFSIG:692] Re: ARLB094 Digital rules clarified (fwd)

From: Johan Forrer <forrerj@ucs.orst.edu>  
Date: Tue, 17 Oct 1995 08:55:05 -0700 (PDT)  
To: hfsig@tapr.org  
Message-ID: <Pine.OSF.3.91.951017083254.21492A-100000@ucs.orst.edu>  
Rendered body type: text/plain

On Mon, 16 Oct 1995, Phil Karn wrote:

> All right!!! Finally we can stop having to defend our stuff against  
> all those complaints of illegality...

>  
> Phil  
>  
>

I'm delighted to see the latest clarification too. At least now there is a way to incorporate modern coding algorithms and improved protocols into digital signaling technology without those rather old and limited restrictions.

What I still hear is that folks seem to think that the signaling waveform is somehow regulated - I do not see anything in the regulations in that regard except perhaps what I do suspect is that they are referring to the kinds of waveform appropriate for the different subbands. That, in my opinion is subject to interpretation, i.e., how to interpret "data", "CW", and "voice". For example, that is perhaps how SSTV ended up in the voice subbands merely by it's modulation designation.

I suppose this one will generate interesting discussions.

We sincerely appreciate the efforts of those that deal with these thorny issues - a thankless job, however, most important in shaping the future of this field.

73's

--Johan, KC7WW

P.S. Those that follow HFSIG, don't be concerned that the traffic volume is so low at present - the list is working quite normally.

---

## 53. Re: [HFSIG:693] Re: ARLB094 Digital rules clarified (fwd)

From: Greg Jones <gjones@tenet.edu>  
Date: Tue, 17 Oct 1995 13:11:53 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199510171811.NAA17390@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

Hello Johan.

The problem still remains that the signalling rate is still in the rules. I talked to Dewayne Hendricks about this last week and we still need to think

about working on these rules once the NPRM on SS rules is taken care of.

Eliminating the signalling rate limits is an important concern.

Cheers - Greg

According to Johan Forrer:

```
>
>
>
> On Mon, 16 Oct 1995, Phil Karn wrote:
>
> > All right!!! Finally we can stop having to defend our stuff against
> > all those complaints of illegality...
> >
> > Phil
> >
> >
>
> I'm delighted to see the latest clarification too. At least now there is
> a way to incorporate modern coding algorithms and improved protocols into
> digital signaling technology without those rather old and limited
> restrictions.
>
> What I still hear is that folks seem to think that the signaling waveform
> is somehow regulated - I do not see anything in the regulations in that
> regard except perhaps what I do suspect is that they are referring to the
> kinds of waveform appropriate for the different subbands. That, in my
> opinion is subject to interpretation, i.e., how to interpret "data",
> "CW", and "voice". For example, that is perhaps how SSTV ended up in
> the voice subbands merely by it's modulation designation.
>
> I suppose this one will generate interesting discussions.
>
> We sincerely appreciate the efforts of those that deal with these thorny
> issues - a thankless job, however, most important in shaping the future of
> this field.
>
> 73's
>
> --Johan, KC7WW
>
> P.S. Those that follow HFSIG, don't be concerned that the traffic volume
> is so low at present - the list is working quite normally.
>
```

---

#### 54. Re: [HFSIG:693] Re: ARLB094 Digital rules clarified (fwd)

From: Mike Curtis <wd6ehr@kaiwan.com>  
Date: Tue, 17 Oct 1995 11:30:40 -0700 (PDT)  
To: hfsig@tapr.org  
Message-ID: <199510171830.LAA01592@kaiwan009.kaiwan.com>  
Rendered body type: text/plain

```
> > All right!!! Finally we can stop having to defend our stuff against
> > all those complaints of illegality...
>
> I'm delighted to see the latest clarification too. At least now there is
> a way to incorporate modern coding algorithms and improved protocols into
> digital signaling technology without those rather old and limited
> restrictions.
>
> What I still hear is that folks seem to think that the signaling waveform
> is somehow regulated - I do not see anything in the regulations in that
> regard except perhaps what I do suspect is that they are referring to the
> kinds of waveform appropriate for the different subbands. That, in my
> opinion is subject to interpretation, i.e., how to interpret "data",
> "CW", and "voice". For example, that is perhaps how SSTV ended up in
> the voice subbands merely by it's modulation designation.
```

I suspect that SSTV ended up in the "voice" subbands because of its bandwidth being "the same" as voice. Also, it's an "attended" mode. I would imagine that wider band data might also be more appropriate in the "voice" subbands

It would also be nice to have the IMHO "artificial" baud rate/data rate restrictions totally eliminated in favor of a specified bandwidth for non-SS systems. Another nicety would be to have access to HF SS on a "minimal interference" basis (it's OK as long as no one squawks about it :-)

-- mike

---

## 55. Re: [HFSIG:694] Re: ARLB094 Digital rules clarified (fwd)

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 17 Oct 1995 11:55:35 -0700  
To: hfsig@tapr.org  
Message-ID: <199510171855.LAA15763@servo.qualcomm.com>  
Rendered body type: text/plain

>Eliminating the signalling rate limits is an important concern.

Right. Here's a classic example of a situation where I'd like to waive these limits in exchange for new limits on power (automatic limiting of Eb/N0)

Phil

---

## 56. Test

From: "Jim Lynch" <k4gvo@america.net>  
Date: Sun, 22 Oct 1995 05:59:48 +0000  
To: hfsig@tapr.org  
Message-ID: <199510221000.KAA26201@america.net>  
Rendered body type: text/plain

Sorry for the wasted bandwidth, but I wanted to see if I was still subscribed to this list. I haven't seen any traffic in a while.

Jim.  
Jim Lynch                      Fayetteville, GA              K4GVO  
'86 GL1200I                  k4gvo@america.net              44.36.34.20

---

## 57. Re: [HFSIG:697] Test

From: lynnc@sos.sos.net  
Date: Sun, 22 Oct 1995 09:16:23 -0700  
To: hfsig@tapr.org  
Message-ID: <199510221616.AA01056@sos.sos.net>  
Rendered body type: text/plain

seems to work  
Lynn

---

## 58. Re: [HFSIG:697] Test

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Sun, 22 Oct 1995 18:53:27 -0800  
To: hfsig@tapr.org  
Message-ID: <9510230151.AA16014@ucs.orst.edu>  
Rendered body type: text/plain

Hi Jim,

Not to worry, all is normal.

I have ordered the parts for the EVM radio interface and will try it out on a Radio Shack 276-147A perf board. The connector catalog is still in the mail, but think there is a Berg connector that will give the right amount of clearance. If all goes well, I will offer a parts kit at my cost is anyone is interested.

Also made a bit of progress on transferring the code to a FLASH Eprom - found a bug in the code as supplied with the EVM that will not allow one to put the modem in EEPROM. A few changes is needed to make all work - I'll post the details soon.

73's

--Johan

---

## 59. PA3DJM experimental telefax interface

From: Reinier van der Lee <rvdlee@austria.cp.philips.com>  
Date: Mon, 23 Oct 1995 10:08:14 +-100  
To: "'TAPR'" <hfsig@tapr.org>  
Message-ID: <01BAA12F.78786180@Reinier>  
Rendered body type: text/plain

In January this year a successful attempt was made to connect two standard office telefax machines via 2 meter ham radio. So far I have not heard of any radio amateur experimenting with telefax. So I guess

this was a "first".

As I moved from Holland to Austria this year, I cannot experiment further with the telefax interface (I passed the exam, but have not yet received an Austrian license). I wrote an article for the Dutch amateur radio magazine "Electron", it will be published later (end this year, early next year).

For amateurs interested in using their fax or fax modem on the VHF/UHF bands, I'm willing to send a description of the schematic diagram. It is not really complicated though. As telefax operates in half-duplex mode, the only thing the interface has to do is separate the in and outgoing signals in interfaces for the transceiver speaker and mike. If your transceiver has no build-in VOX, the PTT-function (push to talk) has to be supplied by the fax interface. All together, I built my experimental design for about \$10,-.

I'm interested in your reply,  
Best regards,

Reinier van der Lee, PA3DJM, Vienna

---

## 60. Re: [HFSIG:700] PA3DJM experimental telefax interface

From: Greg Jones <gjones@tenet.edu>  
Date: Mon, 23 Oct 1995 04:59:20 -0500 (CDT)  
To: hfsig@tapr.org  
Message-ID: <199510230959.EAA22699@Leslie-Francis.tenet.edu>  
Rendered body type: text/plain

There have been several article on fax technology for amateur radio in the PSR over the last few years. Some include:

Oct-90 Issue 40  
9600 bps FAX Test Results Jeff King, WB8WKA  
Test Results of the YamahaV.92 Fax Chip

Oct-90 Issue 40  
FAX Modems and Packet Radio Lyle Johnson, WA7GXD  
An overview of 9600 bps operations using FAX technology on packet radio

Oct-93 Issue 52  
Hi-Speed Digital Communications in Japan Ryuji Suzuki, JF7WEX  
Discussion of packet radio operations in Japan

I think there are some others -- but these are the easiest found.

Cheers - Greg

According to Reinier van der Lee:

```
>
>
> ----- =_NextPart_000_01BAA12F.78818940
> Content-Type: text/plain; charset="iso-8859-1"
> Content-Transfer-Encoding: quoted-printable
>
> In January this year a successful attempt was made to connect two =
> standard office telefax machines via 2 meter ham radio. So far I have =
> not heard of any radio amateur experimenting with telefax. So I guess =
> this was a "first".
>
> As I moved from Holland to Austria this year, I cannot experiment =
> further with the telefax interface (I passed the exam, but have not yet =
> received an Austrian license). I wrote an article for the Dutch amateur =
> radio magazine "Electron", it will be published later (end this year, =
> early next year).=20
>
> For amateurs interested in using their fax or fax modem on the VHF/UHF =
> bands, I=B4m willing to send a description of the schematic diagram. It =
> is not really complicated though. As telefax operates in half-duplex =
> mode, the only thing the interface has to do is separate the in and =
> outgoing signals in interfaces for the transceiver speaker and mike. If =
> your transceiver has no build-in VOX, the PTT-function (push to talk) =
> has to be supplied by the fax interface.=20
> All together, I built my experimental design for about $10,-.
>
> I=B4m interested in your reply,
> Best regards,
>
> Reinier van der Lee, PA3DJM, Vienna
>
>
```

```

> ----- =_NextPart_000_01BAA12F.78818940
> Content-Type: application/ms-tnef
> Content-Transfer-Encoding: base64
>
> eJ8+IhAJAQAQCAEAAAAAAAAABAAEAAQeQBGAIAAAA5AQAAAAAADoAAENGAAQAAgAAAAIAAgABBJAG
> ABgBAAABAAAAADAAAAAADAACAAAAACwAPDgAAAAACaf8PAQAAAF0AAAAAAAATvTvCwCx3EBqhVAgA
> KypWwhUAAAdtQCRVuMjOEajSAIDHUXsQZIMAAAAACBKx+kvqMQGZ1uAN0BD1QCAAAAAFRBUFIa
> U01UUABoZnNpZ0B0YXByLn9yZWAAAAeAAIwAQAAAAUAAABTTVRQAAAAAB4AAzABAAAAADwAAAGhm
> c2lnQHRhcHiub3JnAADABUMAQAAAAAMA/g8GAAAAHgbMAEAAAAHAAAAAJRBUFIInAAACAQswAQAA
> ABQAAABTTVRQ0KHGU01HQFRBUFIuT1JHAAMAAADKAAAAACwBA0gEAAAAACafYPAQAAAAQAAAAAAAC
> mi8BCIAHABgAAABJUE0uTWljcm9zb2Z0IE1haWwuTm90ZQAxCaEEgAEAJgAAAFBBM0RKTsBlEhB1
> cmltZW50YWwgdGVsZWZheCBpbmRlcmZyY2UApw0BBYADAA4AAADLBwoAFwAKAAgADgABABQBASCA
> AwA0AAAAywcKABcACgAGABMAAQAAXAQEJgAEAIQAAAEJDQ0MwRDZEMUMwRENGMTFCMzVBMDA4MEM3
> NTMxQjEwADUHAQQQBgBoBQAEEgAAAAAsAIwAAAAAAAwAAAAAALACKAAAAAAMANGAAAAAAQAA5
> AKA4fhUnoboBHGBwAAEAAAAAAMAAAUeEzREpNIGV4cGVyaW1lbnRhbCB0ZWxlZmF4IGludGVyZmFj
> ZQAAAAIBcQABAAAFgAAAAAG6oScVbW0NzMINHBHPs1oAgMdTGxAAAAB4AHgwBAAAAAwAAAE1TAAAE
> AB8MAQAABUAAABESUNUQVRJT04vUERTNS92ZGx1ZQAAAAADAAYQGAUwSAMABxCbAAwAAHgAIEAEA
> AABIAAAASUS5KQUSV5VZVehJU1lFQVJBU1VDQ0VTU0ZVTEFUVENUFRXQVNNQURFVE9DT050RUNU
> VFdPU1RBTkRBUkRPRkZJQ0VURUxFRkFYUFDSEl0RVNWSUEyTUVURVJIQU1SQURJTWAAAAACAQkQ
> AQAAAMDAADfAwAAowUAAEXaRnUNXL0m/wAKAQ8CFQKoBesCgwBQAvIJAjBjaArAc2V0MjcGAABd
> AoMyA8UCAHBYQnER4nN0ZW0CgzP3AuQHEWKAfQqACM8J2QKABwqBDbELYG5nMTAZTwBQCwoUUQvx
> IEkDoEqFAHB1CsB5IHRoBAAIHL1CsFhIHN1RGNjB5BzZnUDIGHnAkAT4AUxd2EEIADADbdJGgBv
> IAWgbm4fKAVAXHR3HKAwZASRIJpvDdBpGwAAAGVsDcC8YXgcMRfWc4AHkXYHMD4gEeAHgBPQ
> BcARgG04nIcUGlvLgYAHKAewJUfWEkGQXYccG5vBUDuaBpxHfIaoG4Z8CCTGqCHAMAT0AhwIGV4
> cAZx1weAAJlGgCb8KkaEB52+yDjIXBnClAEERoTHAIasMYiHjARoHQilGqFCoV+QQQgIXAEYCGw
> HfADUiCeSAbwF6Ed8ByRQXUTw0SHIRoILCFhYwBwIeIj2P8hIAhwGhAgISS0IiAedwuA9yARHsAe
> USghcAqwBBa00c8tQ1PQIGAREGJ1IgeHpu8aYAVAFhAbAGkowgORKgXNA6BsHkEAgGUpIOAhcL53
> A2AT0DFyCsAkYgMeoPchIAwXLUJEL/ARcCNHivSxAMBnYXofQSBgRR6g8x0QA2BuIisQJMAkoSlw
> 8y/QHHBwdQJgBAaiIB3zewtgIBIoCfAvIiqnGnFsexnwHPB4MJIKwDKxJ3xG/wWxI1UEIC4DB5AT
> 0B3wC4D+ICoQJHisoScQISEe4AwXh7DBHEgcAIgLTNWSEZCL1U/0C/QHaFzKxFc+CdiNCBwN1I9
> ox1h00L/GrANsAT0IMADoCKBLUIE8P8iICNhhKBCMACwCcAgYDLB/wVAGjEh4hYQB0A6AQWgG8DP
> MKEjC58iCGBnaCDgKFE9HoZvI/EjCtyCIEFsZvgTZHULUCPQPuMrEC1C/wIg0gEaET20LfkRgCYh
> HKD+ZBygGjERSAqxI3FKZSKhMx3xL/BnbyRyAJBnbv8HQEgTLgcEIDQWKjAACTETZQXAcYpWYwsg
> ISmSbe5pUNAYwSKQeQhht8tLui8h4C/RAXAd8C09UVZPQlhJZFBUVc0bQG57HRBCwig30DggHIIB
> kGzcaylLRjehGtBwReEo0d5iGfIz8S3a0tdBN3EckL5nEcAssSsSU2IFQG0Z8H8j2AdAQJJOATQD
> KBNNYSdyJBfgLC0nbUC1PKxR49MWEAtQeSwKhUI82TDh7zXwCyBAGCd8UjEgAwAgIWZ2A5EEgSBM
> CeArEFCgQTNEsk0rEFYIkH0c4GEnfBgVGTfKxRUxAAFniAADABAQAAAAAAMAERAAAAAAQAHHMACK
> TNEmobaBQAAIMACKTNEmobaBHGA9AAEAAAAABAAAAAAAGeP
>
> ----- =_NextPart_000_01BAA12F.78818940--
>
>
>

```

## 61. Re: PA3DJM experimental telefax interface

```

From: Peter Schulze <72253.2602@compuserve.com>
Date: 25 Oct 95 05:21:26 EDT
To: HFSIG <hfsig@tapr.org>
Message-ID: <951025092126_72253.2602_EHJ38-4@CompuServe.COM>
Rendered body type: text/plain

> In January this year a successful attempt was made to connect two =
> standard office telefax machines via 2 meter ham radio. So far I have =
> not heard of any radio amateur experimenting with telefax. So I guess =
> this was a "first".

```

Yaesu sells since long a unit called 'FX-1 Radio Fax interface' that comes plug and play and connects any G3 Fax machine to any HF, VHF or UHF transceiver.

As Rainer says, G3-Fax is a simplex mode, thus the box just does the switching and separates the phone line to the Mike/speaker. I remember having installed a couple of these on oil platforms back in 1988.. works fine, even on HF-SSB if the link is strong. Only problem i remember was the DTMF dialling on HF due to transceiver drift.

73 Peter TY1PS

```

>eJ8+IhAJAQAQCAEAAAAAAAAABAAEAAQeQBGAIAAAA5AQAAAAAADoAAENGAAQAAgAAAAIAAgABBJAG
> c2lnQHRhcHiub3JnAADABUMAQAAAAAMA/g8GAAAAHgbMAEAAAAHAAAAAJRBUFIInAAACAQswAQAA...

```

?? what is that ^^ ? chinese ? Sorry i am not a very serious Interneter looks like some coding or compression who can tell me ?

## 62. EVM56K Help Needed

```

From: tomob@ix.netcom.com
Date: Fri, 27 Oct 95 21:13:39 GMT
To: HFSIG@tapr.org (HFSIG)
Message-ID: <M.102795.161339.74@ix-chi11-15.ix.netcom.com>

```



Rendered body type: text/plain

Hello All -

I have been working (although slowly) on finishing the small circuit for the EVM56002 radio interface. Now that I have finished the task, I can't seem to compile/link Pawel's code correctly. I get one error. It refers to an include file called "newqpsk.dat". This file was not available in the zipped package I got from the HFSIG FTP Site. Any help to get me going in the right direction would be appreciated. Thanks in advance.

73 de Tom, N9GUN  
tomob@ix.netcom.com

---

### 63. Re: [HFSIG:699] Re: Test

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sat, 28 Oct 1995 06:44:30 +0200  
To: hfsig@tapr.org  
Message-ID: <199510280444.GAA27897@f15.pix.za>  
Rendered body type: text/plain

Hi Johan

>Also made a bit of progress on transferring the code to a FLASH Eprom -  
>found a bug in the code as supplied with the EVM that will not allow one to  
>put the modem in EEPROM. A few changes is needed to make all work - I'll  
>post the details soon.  
>

Dit klink interessant. Ek het nog nie kans gehad om kode ROMable to maak nie.  
Sal graag wil sien wat jy gedoen het.

Ek het die volgende probleem op die EVM opgetel: ek het 'n toets program geskryf om die IRQA, IRQAB en NMI te toets. Hulle werk almal baie mooi maar ek moes die inset seine vanaf 'n ander PCB per ribbon draad stuur. Baie keer as IRQA,B gestuur word dan kry die uP sommer 'n NMI ook. Ek sien toe op die scope dat daar baie fyn spikes oor koppel vanaf die IRQA/B lyne na die NMI lyn. Volgens die 100MHz scope gaan hulle nie eens deur die CMOS threshold nie maar dis genoeg om die uP te herstel elke keer.

Dit wys nou net weer eens hoe vinnig die 56002 is ! Ek het nog nooit sulke probleme op anders mikro's gekry nie. Die oplossing is toe om die drie onderbrekings lyne van mekaar te skerm en die lyne aan beide kante op te trek weer met 10k weerstande. Mens kan selfs ook klein caps insit om die spikes dood te slaan.

Net gedog ek se jou. Die h/w interrupts werk andersins baie oulik. Ek het ook al die 56002 se unieke event timer gebruik om DSP somme te 'sequence' ipv die SSI poort. Dit werk ook baie mooi. Ek het sodoende 'n afsonderlike heartbeat geskep wat tesame met 'n WDT sorg dat die uP nie ophang nie. Na my probleme met die IRQ's is ek nou baie vir EMI. Die mikro is baie vinnig en sien enige vinnige spike maklik raak op sy beheer lyne. 'n WDT is dus noodsaaklik. Ek gaan moontlik die ADM696 WDT op my PCB bysit gekoppel aan die NMI.

73 Danie

---

### 64. Re: [HFSIG:703] EVM56K Help Needed

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 27 Oct 1995 21:53:06 -0800  
To: hfsig@tapr.org  
Message-ID: <9510280450.AA22820@ucs.orst.edu>  
Rendered body type: text/plain

Hello Tom,

Sorry - my oversight. I have uploaded a new version - EVM56K1D.ZIP. All that is different is the .ZIP file for NEWQPSK in the "SP9VRC" directory that contains all the files that you need to build Pawel's parallel modem.

I will be posting a note shortly on how to program you own flash EEPROM, but will have to lay low for the next few weeks do to exams - thanks for your patience.

Good luck with the experiments.

--Johan, KC7WW

>Hello All -  
>  
>I have been working (although slowly) on finishing the small circuit for the  
>EVM56002 radio interface. Now that I have finished the task, I can't seem to  
>compile/link Pawel's code correctly. I get one error. It refers to an  
include  
>file called "newqpsk.dat". This file was not available in the zipped  
package I got  
>from the HFSIG FTP Site. Any help to get me going in the right direction  
would be  
>appreciated. Thanks in advance.  
>  
>73 de Tom, N9GUN  
>tomob@ix.netcom.com  
>  
>  
>

---

## 65. Re: [HFSIG:703] EVM56K Help Needed

From: JALOCHA@chopin.ifj.edu.pl  
Date: Sat, 28 Oct 1995 18:09 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <E8FC9454A0C057C4@chopin.ifj.edu.pl>  
Rendered body type: text/plain

Have you got the "newqpsk.bas" ? If so, run it with the QBASIC  
and it will make the "newqpsk.dat". If not, I should send you  
the complete set of files...  
Pawel

---

## 66. Re: [HFSIG:706] Re: EVM56K Help Needed

From: tomob@ix.netcom.com  
Date: Sat, 28 Oct 95 17:39:45 GMT  
To: hfsig@tapr.org  
Message-ID: <M.102895.123945.46@ix-pal-ill-21.ix.netcom.com>  
Rendered body type: text/plain

> Have you got the "newqpsk.bas" ? If so, run it with the QBASIC  
> and it will make the "newqpsk.dat". If not, I should send you  
> the complete set of files...  
> Pawel  
>  
>

Hi Pawel -

I don't have all the files yet. If you could send them to me or upload them on  
the HFSIG FTP site, then I can look thru them. With the latest zip file Johan  
just uploaded, I think it will get me started.

I expect to be up and on the air in the next week or so. At that time, I'll  
try and schedule some on-air tests.

73 de Tom, N9GUN  
tomob@ix.netcom.com

---

## 67. Re: [HFSIG:703] EVM56K Help Needed

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sun, 29 Oct 1995 07:41:44 +0200  
To: hfsig@tapr.org  
Message-ID: <199510290541.HAA12463@f15.pix.za>  
Rendered body type: text/plain

Here it is...73 Danie zs6awk

>Hello All -  
>  
>I have been working (although slowly) on finishing the small circuit for the  
>EVM56002 radio interface. Now that I have finished the task, I can't seem to  
>compile/link Pawel's code correctly. I get one error. It refers to an  
include  
>file called "newqpsk.dat". This file was not available in the zipped  
package I got  
>from the HFSIG FTP Site. Any help to get me going in the right direction  
would be  
>appreciated. Thanks in advance.  
>  
>73 de Tom, N9GUN

>tomob@ix.netcom.com  
>  
>  
>  
[Attachment: NEWQPSK.DAT; text/plain; 14,070 bytes]

---

## 68. FW: Re[4]: [HFSIG:700] PA3DJM experimental telefax interface

From: Reinier van der Lee <rvdlee@austria.cp.philips.com>  
Date: Tue, 31 Oct 1995 16:02:53 +100  
To: "'TAPR'" <hfsig@tapr.org>  
Message-ID: <01BAA7AA.5D35DD40@Reinier>  
Rendered body type: text/plain

-----  
From: Marek Kotas[SMTP:mkotas@ccmail.crosscomm.com]  
Sent: Dienstag, 31. Oktober 1995 15:36  
To: rvdlee@austria.cp.philips.com  
Subject: Re[4]: [HFSIG:700] PA3DJM experimental telefax interface

Dear Reinier,

Your files are accessible for anonymous ftp users in  
ftp://cal003109.student.utwente.nl/pub/faxif/ directory. Feel free to  
inform anybody. The Powerpoint presentation is very fine!

Marek,

----- Reply Separator -----  
Subject: RE: Re[2]: [HFSIG:700] PA3DJM experimental telefax interface  
Author: rvdlee@austria.cp.philips.com at inet  
Date: 10/31/95 7:02 AM

Dear Marek,  
I subitted the fax interface description to this FTP site. Two files =  
should be available: Faxif.gif (diagram) and Presfax.ppt (dutch =  
presentation about the interface and backgrounds).  
Thanks for your help sofar.

Reinier

-----

---

# November 1995

Original source file: hfsig.9511

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-11/TAPR-HF\\_1995-11.txt](https://archive.org/download/TAPR-HF_1995-11/TAPR-HF_1995-11.txt)

## 1. mu-Law compandor

From: "Sadowski, P MAJ JCSAR/JT&E" <sadowskp@usafcrqs.nellis.af.mil>  
Date: Mon, 06 Nov 95 08:46:00 cst  
To: hfsig <hfsig@tapr.org>  
Message-ID: <309E91B8@mailgtwy.nellis.af.mil>  
Rendered body type: text/plain

I have a question for the DSP types here. Analog Devices has a neat little evaluation module with dual channel "CD ROM" quality A/D (14 bits resolution) and D/A (also 14 bits resolution)... the sampling rates are up to 44 khz on each channel... and in fact the way the A/D's and D/A's work they do up to 64x oversampling... that's a lot of samples but not necessary for modem work I'm sure...

Anyway, the question I have is based on the fact that the next stage (which can't be bypassed) does an A-Law or mu-Law compandor (nonlinear companding used by telco) so that the output values are no longer 14 bits (nor input values 14 bits) but rather they are 8 bits companded...

Finally :) the question is... what effect does using companded data have on the DSP algorithms? Do you have to modify the values - or just ignore them? And, for modems that rely on AM... what impact does companding have...

Thanks in advance

AH6LS  
sadowskp@usafcrqs.nellis.af.mil

---

## 2. Re: [HFSIG:710] mu-Law compandor

From: Phil Karn <karn@qualcomm.com>  
Date: Mon, 6 Nov 1995 22:30:53 -0800 (PST)  
To: hfsig@tapr.org  
Message-ID: <199511070630.WAA13551@servo.qualcomm.com>  
Rendered body type: text/plain

>Finally :) the question is... what effect does using companded data have on  
>the DSP algorithms? Do you have to modify the values - or just ignore them?

It's common for DSP code to use companded codecs -- we do that in our cell phone vocoders. As I understand it, the first thing you do with a companded sample when you get it is to convert it back to 14-bit linear, so that all DSP operations are on linear samples.

Of course, the quantizing distortion is now a function of the signal amplitude and is worse than a 14-bit linear codec, but DSP arithmetic is so much easier to perform on linear samples -- and you get to use cheap codecs.

Phil

---

## 3. subscribe

From: "droesler" <droesler@cacd.rockwell.com>  
Date: Wed, 15 Nov 95 13:08:35 cst  
To: hfsig@tapr.org  
Message-ID: <9510158164.AA816462639@ccmgw1.cacd.rockwell.com>  
Rendered body type: text/plain

Dan Roesler, HFIA  
droesler@cacd.rockwell.com

---

## 4. Checking

From: fperkins@onramp.net  
Date: Sat, 25 Nov 1995 11:39:25 -0600  
To: hfsig@tapr.org  
Message-ID: <199511251739.LAA07731@mailhost.onramp.net>  
Rendered body type: text/plain

Hi,

Just a test to see if the reflector is still active.

---

**5. Re: [HFSIG:713] Checking**

From: Clifford Buttschardt <cbuttsch@slonet.org>  
 Date: Sat, 25 Nov 1995 10:30:16 -0800 (PST)  
 To: fperkins@onramp.net  
 Cc: hfsig@tapr.org  
 Message-ID: <Pine.OSF.3.91.951125102533.30032A-100000@biggulf.callamer.com>  
 Rendered body type: text/plain

Hi Frank. yes the reflector is operating. Glad to see you are still alive and kicking. You might be interested to know that we still are using the old fashioned TRS80 model 100 laptop to steer antennas!! This time we are using GM4JJJ moontrak program for that! The old QEX interface will be used as before. David programmed his PC-8201 using some unknown calls to the model 100 such as LOCATE which we need to work out, but otherwise the old model T still works! 73 Cliff Buttschardt W6HDO

On Sat, 25 Nov 1995 fperkins@onramp.net wrote:

> Hi,  
 >  
 > Just a test to see if the reflector is still active.  
 >  
 > 73 Frank WB5IPM  
 >  
 >

---

**6. Re: [HFSIG:714] Re: Checking**

From: forrerj@ucs.orst.edu (Johan Forrer)  
 Date: Sun, 26 Nov 1995 15:33:29 -0800  
 To: hfsig@tapr.org  
 Message-ID: <9511262331.AA23246@ucs.orst.edu>  
 Rendered body type: text/plain

Hi all,

I apologise for being so quiet on HFSIG. Hopefully I'll be able to spend a bit more time with you in a little while - like after the second week in January. Still in the middle of exams and just getting by one day at a time.

Welcome to Cliff and the other CCW experimentors - I am quite interested to learn more about the PSK experiments. How about giving us a summary Cliff?

Thanks for your patience - 73's

--Johan, KC7WW

---

**7. Re: [HFSIG:715] Re: Checking**

From: Clifford Buttschardt <cbuttsch@slonet.org>  
 Date: Mon, 27 Nov 1995 13:58:33 -0800 (PST)  
 To: hfsig@tapr.org  
 Cc: hfsig@tapr.org  
 Message-ID: <Pine.OSF.3.91.951127135742.8475B-100000@biggulf.callamer.com>  
 Rendered body type: text/plain

Hi again Johan. OK, let me try and make up a file that is fairly short with the essentials of Coherent CW and BPSK for all! 73 W6HDO

On Sun, 26 Nov 1995, Johan Forrer wrote:

> Hi all,  
 >  
 > I apologise for being so quiet on HFSIG. Hopefully I'll be able to spend a  
 > bit more time with you in a little while - like after the second week in  
 > January. Still in the middle of exams and just getting by one day at a time.  
 >  
 > Welcome to Cliff and the other CCW experimentors - I am quite interested to  
 > learn more about the PSK experiments. How about giving us a summary Cliff?  
 >  
 > Thanks for your patience - 73's  
 >  
 > --Johan, KC7WW  
 >  
 >

---

**8. BPSK modem for satellites**

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>  
Date: Mon, 27 Nov 95 23:47:15 MET  
To: hfsig@tapr.org, dsp4@nic.funet.fi  
Message-ID: <784.jalocho@home.ifj.edu.pl>  
Rendered body type: text/plain

Hi all,

I have made up a 400 bps BPSK modem (for A0-13 telemetry) and I plan to scale it to the 1200 bps version. Is there anybody interested out there who could verify its operation and compare it against some other designs ? The modem runs on the EVM56K or on the DSPCARD4.  
I will try to make tests with my radio equipment but my only VHF/UHF antenna is omnidirectional thus the satellite signals are rather weak.

Pawel

---

## 9. Re: [HFSIG:715] Re: Checking

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Tue, 28 Nov 1995 06:15:26 +0200  
To: hfsig@tapr.org  
Message-ID: <199511280415.GAA17506@f15.pix.za>  
Rendered body type: text/plain

>  
>Welcome to Cliff and the other CCW experimentors - I am quite interested to  
>learn more about the PSK experiments. How about giving us a summary Cliff?  
>

Yes me too !

Danie zs6awk  
danie.brynard@pixie.co.za

---

## 10. Re: [HFSIG:716] Re: Checking

From: "Tony Godshall via aloha.com" <togo@aloha.com>  
Date: Mon, 27 Nov 1995 18:42:42 -1000 (HST)  
To: hfsig@tapr.org  
Message-ID: <Pine.BSD/.3.91.951127182850.24331A-100000@aloha.com>  
Rendered body type: text/plain

On Mon, 27 Nov 1995, Clifford Buttschardt wrote:

> Hi again Johan. OK, let me try and make up a file that is fairly short  
> with the essentials of Coherent CW and BPSK for all! 73 W6HDO

Please do! I anxiously await.

> > Welcome to Cliff and the other CCW experimentors - I am quite interested to  
> > learn more about the PSK experiments. How about giving us a summary Cliff?

I am also! Please forgive me taking up the bandwidth with "Me too" messages, but there seemed to be lots of wasted bandwidth already (all quiet) and I suppose I wanted to de-lurk and let you all know I was out here.

I have some good background with digital algorithms (data correction and compression) and am interested in learning more about radio (my level in tech class) and digital encoding.

My personal idea of a good way to implement digital on HF is to get a good fast DAC and ADC and use a DSP or the main CPU to generate the signal waveform and decode it. That way we could start with a simple encoding system now and move to better systems later with simple upgrades in software. Essentially the software ought to be able to do the following now ...

1. Create transmittable waveforms that represent data
2. Decode such waveforms received.

This ought to be enough to build point to point systems for testing.

As our experience grows, our software could

3. Detect multipath interference and use echo-cancellation techniques to remove it.
4. Perform some kind of bandwidth allocation (CSMA/CD if practical,

TDMA, reservation, what have you) to handle systems more complex than point to point.

Anyway, I digress...

73 de wh6zd

---

## 11. Re: [HFSIG:719] Re: Checking

From: lynnc@sos.sos.net  
Date: Mon, 27 Nov 1995 21:42:49 -0800  
To: hfsig@tapr.org  
Message-ID: <199511280542.AA08506@sos.sos.net>  
Rendered body type: text/plain

Just feel compelled to inject,  
Hey, yr not pulling my (our) leg(s) are u? it sur  
sounds like a high-tech description of what we in the  
ol days called CW (just kidding, I am really absolutely  
fascinated with the mail here)  
Lurker Lynn (W7LTQ) 73

---

## 12. BPSK modem for 1200 bps

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>  
Date: Wed, 29 Nov 95 15:24:45 MET  
To: dsp4@nic.funet.fi, hfsig@tapr.org  
Message-ID: <1244.jalocho@home.ifj.edu.pl>  
Rendered body type: text/plain

Scaling my BPSK modem for 1200 bps did not take that long  
and it's ready now. I could make it lock onto a piece of W0-18  
transmission which I have recorded with my soundcard.  
I will hunt for more of course and try to decode some AX.25 frames.  
Is that possible at all with an omnidirectional antenna ?

For the time being I have few questions regarding the BPSK and satellites:

1. Which filter shape I should use at the demodulator ?  
I have read one G3RUH's article about it and it seems that almost  
every satellite transmits with a different symbol shape ?  
Is there a good compromise or one should really match the shape  
for best noise performance ?
2. Doppler drift: from what I see I should expect up to 80-100 Hz/second  
of carrier drift on 70 cm (and 5 times more on 2400 MHz) ?  
If the tranceiver tunes in 10 Hz steps the UP/DOWN "botton pushing rate"  
gets to some 10 per second which sounds quite many ?  
Is that OK for most tranceivers ?  
What is the typical time needed to hold the button so it reacts ?
3. Transmitting to the satellite: The signal should be 1200 bps  
Manchester encoded which is "almost" the same as 1200 bps BPSK  
on a 1200 Hz carrier. My modem generates BPSK by it's nature  
and there is no problem to tell it to make carrier at 1200 Hz  
but there is no guarantie that the carrier stays synchronous  
with the data stream. Is this the strict requirement for satellite's  
uplink ?

If there is somebody out there who can answer please do.  
Thanks, Pawel.

---

## 13. Re: [HFSIG:719] Re: CCW and BPSK

From: Clifford Buttschardt <cbuttsch@slonet.org>  
Date: Wed, 29 Nov 1995 09:19:39 -0800 (PST)  
To: hfsig@tapr.org  
Cc: Johan Forrer <forrerj@ucs.orst.edu>,  
Bill DeCarle <76170.3362@compuserve.com>  
Message-ID: <Pine.OSF.3.91.951129084942.5285A-100000@biggulp.callamer.com>  
Rendered body type: text/plain

Many thanks to all of you that requested information on Coherent CW and  
the BPSK experiments we are doing. I obtained requests from South Africa  
through Europe and of course from Johan. I must say I am somewhat  
intimidated by this group as we really are doing what might be best  
called the "last hardware breath before DSP!" In fact, the use of the  
Audio Spectrum analyzer in QST for Jan 1992 by Bill DeCarle probably is  
the first step with DSP being conducted by any computer following. Also,  
the Packet Status Register that just arrived certainly indicated that we  
are somewhat reinventing the wheel! Do pay attention to this source as  
well as the twice monthly column in the Digital Journal by Peter-G3IRM.

All of the software advances in our effort is due to VE2IQ, Bill DeCarle in Quebec, Canada, By CC of this message be sure and obtain his information regarding kits, completed units and disks. Start with QST for Jan '92. What you might have read about "sample and hold" techniques are completely out of style now and the best reference for this is QEX magazine for February 1994--"A DSP Version of Coherent-CW (CCW)". This is a practical, real explanation of the technique.

What hardware that is being developed has been done by two non-hams interested in point to point computer communications. Max Carter is the prime innovator of very slow BPSK techniques. His article was in the COMMUNICATIONS QUARTERLY for Fall 1990--"Super Narrowband Techniques Equalize Power Inequity on 1750 Meters". John West in Longmont, Colorado is developing a series of PC boards to be made available using the Max Carter design and DeCarle software. WB6RIJ, WD4PLI and I are doing the final design checkout as we find time to help the project.

BPSK (BiPhaseShiftKeying) is a modification of CCW programs added on to DeCarle's library after version 3.2. Where CCW operates at 12 wpm telegraphic code, BPSK operates at one character per second ASCII! The latter is a weak signal technique developed for the experimenters in the 1750 meter band (160-190 KHz) which is not available all over the world. By the way, comments in PSR by RENALDO, finally elude to BPSK being legal in the ham bands where there was some doubt previously.

While there is so much more to be said about this let me conclude here with at least enough to get the HFSIG interested in more! As you can appreciate our efforts do not replace yours! Cliff Buttschardt W6HDO

---

#### 14. Re: [HFSIG:721] BPSK modem for 1200 bps

From: Clifford Buttschardt <cbuttsch@slonet.org>  
Date: Wed, 29 Nov 1995 10:14:08 -0800 (PST)  
To: hfsig@tapr.org  
Cc: hfsig@tapr.org  
Message-ID: <Pine.OSF.3.91.951129101136.5285H-100000@biggulpcallamer.com>  
Rendered body type: text/plain

Pawel, your comments regarding 1200 baud BPSK are very valuable. We have used the TAPR modem on 300 baud HF but not 1200 baud (except for some brief experiments on ten meters). The questions you ask are in need of answers by all! 73 Cliff Buttschardt W6HDO

---

#### 15. 1200 bps BPSK modem works !

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>  
Date: Wed, 29 Nov 95 22:08:57 MET  
To: hfsig@tapr.org, dsp4@nic.funet.fi  
Message-ID: <793.jalocho@home.ifj.edu.pl>  
Rendered body type: text/plain

I just learned that my new modem works !

I set up my equipment for a W0-18 pass (437.102) - the modem locked onto the signal and I was following the Doppler shift by hand while watching the UP/DOWN LEDs of the DSPCARD4. The lock was being lost few times when the signal was fading completely into noise. It didn't decode any frame...

But very soon there was a pass of L0-19 and I switched to 437.125, the signal sounded stronger, I concentrated on good Doppler tracking and in the middle of the pass I noticed that some 1250 characters have been received on the serial port which meant there were good AX.25 frames !  
The lock would still be lost about ones per minute when the signal sounded like being gone completely but it was coming back afterwards.  
I received in total about 25 frames and 1315 bytes on the KISS port for the whole pass. Some frames were long like these (JN05 trace output):

```
Wed Nov 29 21:47:53 1995 - tnc rcv:
KISS: Port 0 Data
AX25: LUSAT-11->QST-1 UI pid=0xbb
0000 .ZG...p....r.0..Q8a.....v..0...W.....0p..m.....
0040 0W.jy.....0..q,1.AG....@....
```

```
Wed Nov 29 21:48:02 1995 - tnc rcv:
KISS: Port 0 Data
AX25: LUSAT-11->QST-1 UI pid=0xbb
0000 .[G.....EptA]p.S..D..N.-.R...2....#*...w2j..NA..t.4..E&..E.j
0040 .PK.....x{.....S.....44FR1127.8YDPK...
0080 x{.0l.y.....SATPK.....k...7...
00c0 ..~.
```

```
Wed Nov 29 21:48:06 1995 - tnc rcv:
KISS: Port 0 Data
AX25: LUSAT-1->TLM UI pid=Text
```



```
0000 ...0.d...h.m.A...4...d.....4.....{....
0040 .%.Y {!x".#,$(%,&.'.(.).*+...-G../.0.1.2.3I4.5.6.7.8.9.:.;<.
```

My radio setup:

- Discone antenna on the roof
- 25 meters of thin coax (I know, that's terrible for 70cm)
- ICOM R7000 which tunes only in 100 Hz step and it's PLL sounds like it has lot of phase noise. But the lock would not be lost when the receiver was jumping by 100 Hz.

How much luck did I have so I could copy L0-19 ? :-)

Pawel

---

## 16. Re: BPSK modem for 1200 bps

From: rob@pe1chl.ampr.org (Rob Janssen)  
Date: Thu, 30 Nov 1995 12:23:38 +0100 (MET)  
To: jalocha@home-gate.ifj.edu.pl  
Cc: dsp4@nic.funet.fi, hfsg@tapr.org  
Message-ID: <MAA05868@sys3.pe1chl.ampr.org>  
Rendered body type: text/plain

According to Pawel Jalocha:

- > 1. Which filter shape I should use at the demodulator ?
- > I have read one G3RUH's article about it and it seems that almost
- > every satellite transmits with a different symbol shape ?
- > Is there a good compromise or one should really match the shape
- > for best noise performance ?

There are quite some differences between the satellites.

For example, JAS has (had? is it still active?) a very clean signal, while on PACSAT etc there is a lot more phase noise. This was a problem with the G3RUH modem which had an unsuitable filter for it and needed to be modified.

Also, some of the satellites have two different transmitters: the standard one and a "raised cosine" transmitter. I think one of the two has failed on one or more sats, so you find a mixture of signal types.

With your marginal signal you may be in the position to see difference between the different decoding techniques.

- > 2. Doppler drift: from what I see I should expect up to 80-100 Hz/second
- > of carrier drift on 70 cm (and 5 times more on 2400 MHz) ?
- > If the transceiver tunes in 10 Hz steps the UP/DOWN "bottom pushing rate"
- > gets to some 10 per second which sounds quite many ?
- > Is that OK for most transceivers ?
- > What is the typical time needed to hold the button so it reacts ?

This varies widely between transmitters...

Some may be able to do it, some not. This really needs to be configurable. (see my other mail)

It is also possible to send CAT commands to many transmitters, for a absolute frequency update. It could be a possibility to use that to send updates at a certain rate, where the step is different each time. Unfortunately, it is much more complicated, and even more dependent on the transceiver make and model...

Also, absolute frequency control is more suitable for a tracker type program. Maybe it should work in co-operation: tracker sends absolute frequencies, modem informs tracker of error. When the keys and station time are reasonably accurate, the error should be quite constant and only dependent on temperature etc.

I have used open-loop control (tracker controls the frequency, no report back from modem) on 9600 bps, but it is difficult on 1200 bps because of the extra needed accuracy.

- > 3. Transmitting to the satellite: The signal should be 1200 bps
- > Manchester encoded which is "almost" the same as 1200 bps BPSK
- > on a 1200 Hz carrier. My modem generates BPSK by it's nature
- > and there is no problem to tell it to make carrier at 1200 Hz
- > but there is no guarantee that the carrier stays synchronous
- > with the data stream. Is this the strict requirement for satellite's
- > uplink ?

I think so... but I don't know for sure.

The original G3RUH modem just uses a XOR between data and clock,

so it is synchronous. You could do the same (with a lowpass filter after it), I think it should be an easy addition.

Rob

```
--
+-----+-----+-----+-----+-----+-----+
| Rob Janssen rob@knoware.nl | BBS: +31-302870036 (2300-0730 local) |
| AMPRnet: rob@pelchl.ampr.org | AX.25 BBS: PE1CHL@PI8WNO.#UTR.NLD.EU |
+-----+-----+-----+-----+-----+-----+

```

---

## 17. Loading problems with EVM56

From: tomob@ix.netcom.com  
Date: Thu, 30 Nov 95 19:12:52 GMT  
To: HFSIG@tapr.org (HFSIG)  
Message-ID: <M.113095.131252.06@ix-chi10-03.ix.netcom.com>  
Rendered body type: text/plain

Hello All -

I have completed the building of the interface circuit for the EVM56. I believe I am having trouble in my procedures to load the code needed to run the multi-tone modem. I first load BIOS.CLD via the debugger, and then try to load the modem software. At that time, the debugger crashes and locks up the entire PC forcing me to reset the PC and start over again. What am I doing wrong????? Please help!!!

73 de Tom, N9GUN  
tomob@ix.netcom.com

---

## 18. Re: [HFSIG:721] BPSK modem for 1200 bps

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Fri, 1 Dec 1995 07:18:41 +0200  
To: hfsig@tapr.org  
Message-ID: <199512010518.HAA19703@f15.pix.za>  
Rendered body type: text/plain

Hi All

First of all I am also learning. So don't think everything I say is true :-)

>Scaling my BPSK modem for 1200 bps did not take that long  
>and it's ready now. I could make it lock onto a piece of W0-18  
>transmission which I have recorded with my soundcard.  
>I will hunt for more of course and try to decode some AX.25 frames.

Webersat 18 has a 1200bps BPSK (on SSB) AX.25 uplink on 437.075MHz. It uses a Raised Cosine 1200bps BPSK (on SSB) AX.25 downlink on 437.102MHz. In other words this is a PSK packet radio transponder. Very interesting...It also has a ATV NTSC camera on board. We in ZS however uses PAL :-)

>1. Which filter shape I should use at the demodulator ?  
> I have read one G3RUH's article about it and it seems that almost  
> every satellite transmits with a different symbol shape ?  
> Is there a good compromise or one should really match the shape  
> for best noise performance ?

So Pawel W0-18 uses Raised Cosine on downlink.

The 400bps beacon on A0-13 is however differentially encoded and then Manchester encoded by XORing the data by the clock of 400Hz. A third order Bessel at 560Hz LPF is then applied. This resulting signal is then fed directly into a balanced modulator ie like a SSB transmitter. So please correct me if I am wrong but this is different to normal 2m FM packet radio. They use NRZ-S encoding on 2m packet so to speak.

Can a 400bps BPSK A0-13 beacon decoded datastream be fed into a 2m packet HDLC decoder ?

>  
>2. Doppler drift: from what I see I should expect up to 80-100 Hz/second  
> of carrier drift on 70 cm (and 5 times more on 2400 MHz) ?  
> If the transceiver tunes in 10 Hz steps the UP/DOWN "bottom pushing rate"  
> gets to some 10 per second which sounds quite many ?  
> Is that OK for most transceivers ?  
> What is the typical time needed to hold the button so it reacts

Yes I am using an IC-275. It has a datamode so its synthesizer can switch

very rapidly between tx and rx and rx and rx. And think it is done in about 5ms. So ten times a second is 100ms which is 10x slower.

?

>

>3. Transmitting to the satellite: The signal should be 1200 bps  
> Manchester encoded which is "almost" the same as 1200 bps BPSK  
> on a 1200 Hz carrier. My modem generates BPSK by it's nature  
> and there is no problem to tell it to make carrier at 1200 Hz  
> but there is no guarantee that the carrier stays synchronous  
> with the data stream. Is this the strict requirement for satellite's  
> uplink ?

Yes here again I need some confirmation. As I understand it they use NRZ-S line coding on AX.25 but this is done in the HDLC part of the software ?

I am also wondering about the 'carrier stays synchronous with the datastream'. Is this really necessary and why ? I see that the literature for some 1200bps PSK modems say that there is a jumper where you can apply your own clock if you want to create a Manchester stream at a different frequency. I was also wondering if this clock must have a certain phase relationship with the datastream. This I can say: on the JAS-1/F0-12 PSK modem there is TP4 where one can apply your own 1200Hz clock. I don't know whether this satellite is still active but that is what its docs say. May be other 1200bps PSK sats are similar.

Hope there is some expert out there who can confirm/deny/explain all this.

73 Danie ZS6AWK

---

# December 1995

Original source file: hfsig.9512

Source: <https://archive.org/details/tapr-1997>

Recovered from: [https://archive.org/download/TAPR-HF\\_1995-12/TAPR-HF\\_1995-12.txt](https://archive.org/download/TAPR-HF_1995-12/TAPR-HF_1995-12.txt)

## 1. Re: [HFSIG:726] Loading problems with EVM56

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Thu, 30 Nov 1995 22:35:58 -0800  
To: hfsig@tapr.org  
Message-ID: <9512010633.AA05633@ucs.orst.edu>  
Rendered body type: text/plain

Tom,

>From what you describe it sounds like you have your COM port for the EVM not set up right. Most PC's really only have two com port IRQ's (3 and 5), where COM1/3 share IRQ 3 and COM2/4 share IRQ 5. If you connected your EVM to COM3 for instance and then have your mouse connected to COM1, then the debugger will grab IRQ 3 and use it. So when the OnCe port talks to the PC, not only does the debugger receive those interrupts, but the mouse's device driver will also receive those interrupts - its the mouse driver that eventually choke on that data. So - if you can check that to make sure. You need a dedicated COM port with its own free IRQ. I know that is a scarce resource.

Hope that helps. Hope to be doing some tests with you in the near future.

--Johan

>Hello All -

>

>I have completed the building of the interface circuit for the EVM56. I believe I

>am having trouble in my procedures to load the code needed to run the multi-tone

>modem. I first load BIOS.CLD via the debugger, and then try to load the modem

>software. At that time, the debugger crashes and locks up the entire PC forcing me

>to reset the PC and start over again. What am I doing wrong????? Please help!!!

>

>73 de Tom, N9GUN

>tomob@ix.netcom.com

>

>

>

---

## 2. Re: [HFSIG:728] Re: Loading problems with EVM56

From: David Wilson Borden <dwborden@ns1.pananet.com>

Date: Fri, 01 Dec 1995 06:21:34 -0500

To: hfsig@tapr.org

Message-ID: <30BEE53E.210C@pananet.com>

Rendered body type: text/plain

John,

I am trying to buy an EVM56 and join you people. I am going through Newark Electronics. I am down here in Panama, but that should not matter, I have the U.S. Mail through a military base, so it should eventually work.

As a member of AMRAD, we build 56001 boards several years ago, but we never made them in quantity and we wire wrapped them, so I am looking to move to something more standard, and the EVM56 seems cheapest.

On your interrupts, although for your discussion it does not matter, since you are talking about conflicts:

INT 0 = System Timer

INT 1 = Keyboard

INT 2 = System

INT 3 = COM2 (Usually modem or I use to load my 56001 board)

INT 4 = COM1 (Usually mouse as you point out)

INT 5 = Free (I use for ethernet card to talk to other PCs)

INT 6 = Disk Controller

INT 7 = Free (I use for sound card like SoundBlaster)

Looking forward to getting the board and trying out all this neat new software.....David, K8MM0/HP2

### 3. 1200 bps BPSK modem works !

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>  
Date: Fri, 1 Dec 95 13:52:22 MET  
To: hfsig@tapr.org, dsp4@nic.funet.fi  
Message-ID: <1262.jalocho@home.ifj.edu.pl>  
Rendered body type: text/plain

I got a confirmation from an "independent source", Pauli Jeskanen, that my 1200 bps modem works well in his "real" satellite setup. At least on receiving but he is going to try transmissions as well. If that works we have got a satellite modem on the DSPCARD4 and EVM56K !

For the 400 bps modem for the telemetry, I have to find out the details about the data encoding and try again. My naive assumption that it's just like the 1200 bps downlinks did not work :-)

Still you can use this modem for HF...  
Actually I could scale it to some very low bit rate (10 bps ?) for (super) narrow band experiments.

Pawel

### 4. Negative Frequencies

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)  
Date: Fri, 1 Dec 95 07:32:45 CST  
To: hfsig@tapr.org  
Cc: mcdermot@rockdal.aud.alcatel.com  
Message-ID: <9512011332.AA01784@eagle.aud.alcatel.com>  
Rendered body type: text/plain

Danie: I think the email I sent you previously on negative frequencies was not so clear. I had in my mind some FFT frequency wrapping effects, and not purely the subject of negative frequencies, for some reason.

I have reread John Bloom (KE3Z) article in Sept 94 QEX, and it is very clear. You should read this, as I think it will explain better what I think your question was.

To be concise: if you have a signal, then to get rid of it's negative part, you must have the I- and Q- components. In analog you use a 90-degree wideband phase shifter (or plus 45 and minus 45-degree shifters). In the digital world, you can use a Hilbert transform to generate 90-degree phase shift. You end up with two quadrature signals, the I- and Q- components, which unambiguously defines the positive (or negative, with a sign change) frequency parts.

I hope this is a better answer, if not please let me know.

- Tom

|                                        |                            |
|----------------------------------------|----------------------------|
| -----+-----                            |                            |
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |
| -----+-----                            |                            |

### 5. Re: [HFSIG:726] Loading problems with EVM56

From: JALOCHA@chopin.ifj.edu.pl  
Date: Fri, 1 Dec 1995 15:43 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <8C3F4B9FC020951F@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>I have completed the building of the interface circuit for the EVM56. I believe I  
>am having trouble in my procedures to load the code needed to run the multi-tone  
>modem. I first load BIOS.CLD via the debugger, and then try to load the modem  
>software. At that time, the debugger crashes and locks up the entire PC forcing me  
>to reset the PC and start over again. What am I doing wrong????? Please help!!!

Hello Tom, do other applications like FFT-CUT or COREFLT work correctly ?  
Just check maybe you didn't set the EVM56K flag in the NEWQPSK.ASM  
before compiling ? I have no idea why it would crash the PC but...  
If you compiled the source for the DSPCARD4 the only difference  
is in the data memory allocation, thus some blocks of data go into

different places of the DSP's RAM. Maybe this touches some vital locations ? And check if the memory jumper is set to "16K".

Pawel

---

## 6. Re: [HFSIG:727] Re: BPSK modem for 1200 bps

From: Alan Bloom <alanb@tsnake2.sr.hp.com>  
Date: Fri, 1 Dec 1995 09:33:14 -0800 (PST)  
To: hfsig@tapr.org (Danie Brynard)  
Message-ID: <199512011733.AA046749195@tsnake2.sr.hp.com>  
Rendered body type: text/plain

>

> Webersat 18 has a 1200bps BPSK (on SSB) AX.25 uplink on 437.075MHz. It uses  
> a Raised Cosine 1200bps BPSK (on SSB) AX.25 downlink on 437.102MHz.

Is it raised cosine or root raised cosine? Generally a root Nyquist filter is used in both receiver and transmitter, so that the total channel response is Nyquist, which gives zero inter-symbol interference (ISI).

Also, you need to know the alpha factor. Transmitter and receiver must use the same alpha factor to avoid ISI.

Alan Bloom N1AL

---

## 7. Re: [HFSIG:729] Re: Loading problems with EVM56

From: Alan Bloom <alanb@tsnake2.sr.hp.com>  
Date: Fri, 1 Dec 1995 09:37:57 -0800 (PST)  
To: hfsig@tapr.org (David Wilson Borden)  
Message-ID: <199512011737.AA123619477@tsnake2.sr.hp.com>  
Rendered body type: text/plain

>

> INT 0 = System Timer  
> INT 1 = Keyboard  
> INT 2 = System  
> INT 3 = COM2 (Usually modem or I use to load my 56001 board)  
> INT 4 = COM1 (Usually mouse as you point out)  
> INT 5 = Free (I use for ethernet card to talk to other PCs)  
> INT 6 = Disk Controller  
> INT 7 = Free (I use for sound card like SoundBlaster)

Aren't INT5 and INT7 generally used for parallel printer ports LPT2 and LPT1?

Al Bloom N1AL

---

## 8. Re: [HFSIG:728] Re: Loading problems with EVM56

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sat, 2 Dec 1995 07:07:19 +0200  
To: hfsig@tapr.org  
Message-ID: <199512020507.HAA26876@f15.pix.za>  
Rendered body type: text/plain

I use to have endless pains with swopping com ports. Now I have installed another 2 com ports and configured them as com3 & 4 and leave the EVM and its ONCE on them permanently. So I don't have to diconnect the mouse every time. It seem to work OK.

73 danie

>Tom,

>

>>From what you describe it sounds like you have your COM port for the EVM not  
>set up right. Most PC's really only have two com port IRQ's (3 and 5), where  
>COM1/3 share IRQ 3 and COM2/4 share IRQ 5. If you connected your EVM to COM3  
>for instance and then have your mouse connected to COM1, then the debugger  
>will grab IRQ 3 and use it. So when the OnCe port talks to the PC, not only  
>does the debugger receive those interrupts, but the mouse's device driver  
>will also receive those interrupts - its the mouse driver that eventually  
>choke on that data. So - if you can check that to make sure. You need a  
>dedicated COM port with its own free IRQ. I know that is a scarce resource.

>

>Hope that helps. Hope to be doing some tests with you in the near future.

>

>--Johan

>

>

```

>
>>Hello All -
>>
>>I have completed the building of the interface circuit for the EVM56. I
>beieve I
>>am having trouble in my prodcedures to load the code needed to run the
>multi-tone
>>modem. I first load BIOS.CLD via the debugger, and then try to load the
modem
>>software. At that time, the debugger crashes and locks up the entire PC
>forcing me
>>to reset the PC and start over again. What am I doing wrong????? Please
>help!!!
>>
>>73 de Tom, N9GUN
>>tomob@ix.netcom.com
>>
>>
>>
>
>

```

## 9. Re: [HFSIG:731] Negative Frequencies

From: danie.brynard@pixie.co.za (Danie Brynard)  
 Date: Sat, 2 Dec 1995 07:07:34 +0200  
 To: hfsig@tapr.org  
 Message-ID: <199512020507.HAA26885@f15.pix.za>  
 Rendered body type: text/plain

Hi TOM

```

>
> Danie: I think the email I sent you previously on
>negative frequencies was not so clear. I had in my mind
>some FFT frequency wrapping effects, and not purely the subject
>of negative frequecies, for some reason.
>
> I have reread John Bloom (KE3Z) article in Sept 94 QEX,
>and it is very clear. You should read this, as I think it will
>explain better what I think your question was.
>
> To be concise: if you have a signal, then to get rid of
>it's negative part, you must have the I- and Q- components. In
>analog you use a 90-degree wideband phase shifter (or plus 45
>and minus 45-degree shifters). In the digital world, you can
>use a Hilbert transform to generate 90-degree phase shift. You
>end up with two quadrature signals, the I- and Q- components,
>which unambiguously defines the positive (or negative, with a
>sign change) frequency parts.
>

```

Yes but that is the problem, once you have created this complex signal from the real input signal, how do you determine whether that resultant frequency is positive or negative ? It is not just a matter of looking at the sign of the signal, you have to calculate the sign by looking at two successive time samples of the complex signal to see which direction the complex phasor is turning.

I am looking for an elegant and easy way to do this. You can assume that I will have two time samples of the complex signal ie

```

y(t)=re1+jim1
y(t-1)=re2+jim2

```

Now which way is the phasor turning ? Clockwise or anticlockwise ?

73 Danie

```

> I hope this is a better answer, if not please let me know.
>
>
>
>
>

```

- Tom

```

>-----+-----
> Tom McDermott | "All opinions expressed
> Alcatel Network Systems, Inc. | are my own, and do not
> mcdermot@aud.alcatel.com | represent those of Alcatel
> [ICC'96 Technical Program Secretary] | Network Systems, Inc."

```

> [ June 23-27, 1996, Dallas, Tx. ] |  
>-----+-----  
>  
>  
>

---

## 10. Re: [HFSIG:732] Re: Loading problems with EVM56

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sat, 2 Dec 1995 07:07:38 +0200  
To: hfsig@tapr.org  
Message-ID: <199512020507.HAA26890@f15.pix.za>  
Rendered body type: text/plain

Your mouse driver is probably still active and also receiving data from the EVM ?!

danie zs6awk

>>I have completed the building of the interface circuit for the EVM56. I believe I  
>>am having trouble in my procedures to load the code needed to run the multi-tone  
>>modem. I first load BIOS.CLD via the debugger, and then try to load the modem  
>>software. At that time, the debugger crashes and locks up the entire PC forcing me  
>>to reset the PC and start over again. What am I doing wrong????? Please help!!!  
>  
>Hello Tom, do other applications like FFT-CUT or COREFLT work correctly ?  
>Just check maybe you didn't set the EVM56K flag in the NEWQPSK.ASM  
>before compiling ? I have no idea why it would crash the PC but...  
>If you compiled the source for the DSPCARD4 the only difference  
>is in the data memory allocation, thus some blocks of data go into  
>different places of the DSP's RAM. Maybe this touches some vital  
>locations ? And check if the memory jumper is set to "16K".  
>  
>Pawel  
>  
>

---

## 11. Re: [HFSIG:733] Re: BPSK modem for 1200 bps

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Sat, 2 Dec 1995 07:07:45 +0200  
To: hfsig@tapr.org  
Message-ID: <199512020507.HAA26898@f15.pix.za>  
Rendered body type: text/plain

We will have to ask James Miller G3RUH. My documentation says just raised cosine but your are of course right. The root is normally taken.

I will ask him or perhaps James are you seeing this ?

danie zs6awk

>>  
>> Webersat 18 has a 1200bps BPSK (on SSB) AX.25 uplink on 437.075MHz. It uses  
>> a Raised Cosine 1200bps BPSK (on SSB) AX.25 downlink on 437.102MHz.  
>  
>Is it raised cosine or root raised cosine? Generally a root Nyquist  
>filter is used in both receiver and transmitter, so that the total channel  
>response is Nyquist, which gives zero inter-symbol interference (ISI).  
>  
>Also, you need to know the alpha factor. Transmitter and receiver must  
>use the same alpha factor to avoid ISI.  
>  
>Alan Bloom N1AL  
>  
>

---

## 12. Re: [HFSIG:736] Re: Negative Frequencies

From: Phil Karn <karn@qualcomm.com>  
Date: Sun, 03 Dec 1995 21:50:26 -0800  
To: hfsig@tapr.org  
Message-ID: <199512040206.SAA07502@qualcomm.com>  
Rendered body type: text/plain



At 11:24 PM 12/1/95 -0600, you wrote:

```
>y(t)=re1+jim1
>y(t-1)=re2+jim2
>
>Now which way is the phasor turning ? Clockwise or anticlockwise ?
```

Seems pretty straightforward to me. Assuming no aliasing (an important assumption), all you have to do is convert the two samples to polar notation, computing just the angle part since that's all we're interested in here:

```
phi(t) = atan2(im1,re1)
phi(t-1) = atan2(im2,re2)
```

where  $\text{atan2}(y,x) = \arctangent(y/x)$  with automatic quadrant placement. (Many math libraries since Fortran days have this function built in.)

Now you just subtract the two angles mod  $2\pi$ . If one is, say,  $\pi/2$  larger than you can consider the vector to have rotated in that direction. Of course, it's also possible that it rotated  $3\pi/2$  (270 deg) in the opposite direction, which is why it's important to assume no aliasing.

Phil

---

### 13. Re: [HFSIG:725] Re: BPSK modem for 1200 bps

```
From: Phil Karn <karn@qualcomm.com>
Date: Mon, 04 Dec 1995 09:41:00 -0800
To: hfsig@tapr.org
Message-ID: <199512041357.FAA20352@qualcomm.com>
Rendered body type: text/plain
```

At 06:48 AM 11/30/95 -0600, you wrote:

```
>Also, some of the satellites have two different transmitters: the standard
>one and a "raised cosine" transmitter. I think one of the two has failed
>on one or more sats, so you find a mixture of signal types.
```

I've heard that at least one of the Microsats has problems with its PSK modulators that have caused the phase shift to deviate from antipodal (0/180 deg). One effect is to add residual carrier. Another is to totally screw up Costas loops that depend on the signal being antipodal. If you work out what's happening, you can see that the data isn't properly cancelled from the VCO correction signal, causing considerable loop jitter. In fact, for certain phase shifts you can show that the loop doesn't even lock.

A modified demod should be able to deal with this problem. Offhand I'd suggest shifting the phase of the quadrature carrier to something other than 90 deg until the correction signal becomes clean again.

Phil

---

### 14. Re: [HFSIG:738] Re: BPSK modem for 1200 bps

```
From: JALOCHA@chopin.ifj.edu.pl
Date: Mon, 4 Dec 1995 12:20 GMT+1
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>
Message-ID: <CB6A0B4D0020B1C3@chopin.ifj.edu.pl>
Rendered body type: text/plain
```

```
>We will have to ask James Miller G3RUH. My documentation says just raised
>cosine but your are of course right. The root is normally taken.
```

So what is mathematically the root raised cosine ?  
Is it  $\sin(x)/x$  ?

The other thing is that I would like to know which filter to use on the \_receiver\_ side. Suppose the transmitter is raised cosine:  $0.5 + 0.5 \cos(x)$  for  $x = -\pi \dots \pi$  what should be the receiver's shape ? I suspect you need  $\sin(x)/x \dots$  but this one is infinitely long !

Over the weekend I scaled the BPSK modem the other way, down to 30 bps. Then I made up a simple terminal interface:

- pressing Ctrl-T keys the transmitter
- pressing Ctrl-R releases the PTT
- any other character gets transmitted on the carrier as a 7-bit ASCII in asynchronous-like format.
- Any received character is printed on the screen

The signal's bandwidth is some 30 Hz at -6dB and 60 Hz at -50 dB. The specific operating feature is that the receiver sweeps its carrier

within given margins to find the lock and this takes time...  
for +/- 20 Hz sweep the lock may take 5-15 seconds.  
The transmitting station needs to send that much of idle/dummy  
information.

If there is some interest I could go on and add some FEC...

Pawel

---

## 15. Re: Negative Frequencies

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)  
Date: Mon, 4 Dec 95 09:49:31 CST  
To: hfsig@tapr.org  
Message-ID: <9512041549.AA02311@eagle.aud.alcatel.com>  
Rendered body type: text/plain

> Yes but that is the problem, once you have created this complex signal from  
> the real input signal, how do you determine whether that resultant frequency  
> is positive or negative ? It is not just a matter of looking at the sign of  
> the signal, you have to calculate the sign by looking at two successive time  
> samples of the complex signal to see which direction the complex phasor is  
> turning.  
>  
> I am looking for an elegant and easy way to do this. You can assume that I  
> will have two time samples of the complex signal ie  
>  
>  $y(t) = \text{re1} + \text{jim1}$   
>  $y(t-1) = \text{re2} + \text{jim2}$   
>  
> Now which way is the phasor turning ? Clockwise or anticlockwise ?  
>  
> 73 Danie

OK. According to the definition previously, a phasor rotation of  
counter-clockwise is POSITIVE frequency (the imaginary part is 90-degrees  
delayed from the real part). Convention is that zero-degrees is the EAST  
point of the graph.

Now we have to understand the nature of the signals in order  
to develop an algorithm. Let's start with the simplest case - single  
frequency (or very narrow bandpass). Assume that the sampling rate is  
high enough so that we can localize the zero-crossing of the signals.  
Then at the time of the positive-going zero-crossing of the real axis,  
the imaginary will be 90-degrees later, and thus will be a negative voltage  
(when the frequency is positive). If the frequency is negative, then the  
voltage  
would be positive on the imaginary signal at that time.

After this, it starts to get complicated. For a wideband signal,  
each frequency component is delayed by 90-degrees, thus the amount of delay  
is proportional to frequency. So for a wideband signal, the waveform shape of  
the imaginary signal is not so obvious as compared to the real signal. If we  
perform a Hilbert transform on the real data, we will add another 90-degrees  
of  
delay to it, resulting in a total of 180-degrees. If the  
transformed-imaginary  
data is the inverse-sign of the real data, then we know that the signal was  
positive frequency. However, if the transformed-imaginary data is the  
same-sign  
as the real data, then we know that the signal was negative frequency.

I am not sure if there is a simple expression for comparing  
the complex time-domain samples  $y(t)$ ,  $y(t-1)$  to determine the phasor rotation  
direction in the general case. If the signal has been filtered enough, then  
it is possible, but if the signal is fairly wideband, then it is more  
difficult.  
I have not explored this latter case, and do not have a reference to the  
literature.

- Tom, N5EG

---

Tom McDermott  
Alcatel Network Systems, Inc.  
mcdermot@aud.alcatel.com

| "All opinions expressed  
| are my own, and do not  
| represent those of Alcatel

|                                        |                        |
|----------------------------------------|------------------------|
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc." |
| [ June 23-27, 1996, Dallas, Tx. ]      |                        |

----- End Included Message -----

|                                        |                            |
|----------------------------------------|----------------------------|
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |

## 16. Motorola DSP?

From: N10BU@MINDPORT.NET (Douglas M. Pervine)  
 Date: Thu, 29 Oct 1959 09:55:34 GMT  
 To: hfsig@tapr.org  
 Message-ID: <ecdc33b4.471146@MAIL.MINDPORT.NET>  
 Rendered body type: text/plain

I'm looking for some General/Technical information about the Motorola DSP 56002EVM. I've searched Motorola's Home Page and found nothing. Thanks!

--73's  
 V/R DMP

## 17. Re: [HFSIG:741] Re: BPSK modem for 1200 bps

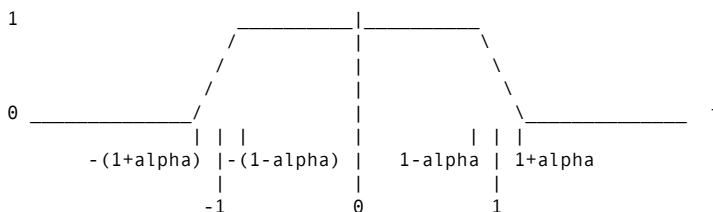
From: Alan Bloom <alanb@tsnake2.sr.hp.com>  
 Date: Mon, 4 Dec 1995 09:51:08 -0800 (PST)  
 To: hfsig@tapr.org  
 Message-ID: <199512041751.AA084949468@tsnake2.sr.hp.com>  
 Rendered body type: text/plain

> So what is mathematically the root raised cosine ?  
 > Is it  $\sin(x)/x$  ?

A (non-root) raised-cosine filter has a frequency response that is (what else) a raised cosine:

$$\begin{aligned}
 P(f) &= 1 & |f| &\leq (1-\alpha) \\
 P(f) &= 0.5 \left( 1 + \cos \left( \frac{\pi (f - (1-\alpha))}{2\alpha} \right) \right) & (1-\alpha) < |f| < (1+\alpha) \\
 P(f) &= 0 & |f| &\geq (1+\alpha)
 \end{aligned}$$

Where the amplitude and the cutoff frequency (half-amplitude point) have been normalized to 1. (The cutoff frequency is at 1/2 the symbol rate.)



The root raised-cosine filter has the square root of that frequency response. That makes the cutoff frequency the half-power point, rather than the half-amplitude point.

The time-domain response of the raised-cosine is a weighted sinc function:

$$p(t) = \text{sinc}(2t) \frac{\cos(2\pi\alpha t)}{1 - 16(\alpha)^2 t^2}$$

where  $\text{sinc}(x) = \sin(\pi x) / (\pi x)$ . Since  $\sin(2\pi t)$  equals zero for  $t = 1, 2, 3, \dots$  the total signal is zero at the decision points of all adjacent symbols. i.e. no inter-symbol interference = Nyquist filter.

The time-response for a ROOT raised-cosine filter is more complicated, and does NOT go to zero at the unwanted desision points. So if you try to receive a root-Nyquist signal without using the complementary root-Nyquist filter in the receiver, you will have severe inter-symbol interference.

If the transmitter uses a (non-root) raised-cosine filter, then the receiver must also use a Nyquist filter of some type to avoid inter-symbol interference.

Both the raised-cosine and root raised-cosine filters theoretically have a time response that goes out to +/- infinity. So of course a practical FIR filter must be truncated. Actually, you usually use some windowing function for the purpose, to minimize ripple in the frequency response. After windowing, a Nyquist filter will remain Nyquist. However, a root-Nyquist filter may show some (usually small) inter-symbol interference, depending on how the windowing is done.

Alan Bloom N1AL

---

## 18. Re: [HFSIG:743] Motorola DSP?

From: Johan Forrer <forrerj@ucs.orst.edu>  
Date: Mon, 4 Dec 1995 11:11:29 -0800 (PST)  
To: hfsig@tapr.org  
Message-ID: <Pine.OSF.3.91.951204110607.10368A-100000@ucs.orst.edu>  
Rendered body type: text/plain

Douglas,

You should check out an article that I wrote about the EVM in QEX (I think August 1995 - The ARRL publishes QEX and you should be able to obtain back issues from them) "Using the Motorola DSP56002 EVM for Amateur Radio DSP". You will find it a useful source of information on various Amateur radio-related projects that you can use the EVM for. Also some technical details of the hardware/software of the EVM.

--Johan Forrer, KC7WW

On Mon, 4 Dec 1995, Douglas M. Pervine wrote:

> I'm looking for some General/Technical information about the  
> Motorola DSP 56002EVM. I've searched Motorola's Home Page and found  
> nothing. Thanks!  
>  
> --73's  
> V/R DMP  
>  
>

---

## 19. Re: [HFSIG:743] Motorola DSP?

From: David Wilson Borden <dwborden@ns1.panenet.com>  
Date: Mon, 04 Dec 1995 19:08:42 -0500  
To: hfsig@tapr.org  
Message-ID: <30C38D8A.7FCB@panenet.com>  
Rendered body type: text/plain

I to followed the Motorola Web page and did not find a good description, just hype about what it does. A really good start is the August 1995 QEX article "Using the Motorola DSP56002EVM for Amateur Radio DSP Projects by Johan Forrer, KC7WW. After you read that, you could download EVM56K1.ZIP from the TAPR Web Page vector to ftp.tapr.org/tapr/SIG/hfsig/upload. I did that and now have a floppy disk full of source code to study. The code is commented well and if you have the 56000 family manual or the User's Manual, you should be able to follow along. I just have to figure out how to read the schematic of the interface circuit I downloaded.....David,K8MMO/HP2

---

## 20. Re: [HFSIG:744] Re: BPSK modem for 1200 bps

From: JALOCHA@chopin.ifj.edu.pl  
Date: Tue, 5 Dec 1995 13:32 GMT+1  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-ID: <9E97B07D6020C543@chopin.ifj.edu.pl>  
Rendered body type: text/plain

>A (non-root) raised-cosine filter has a frequency response that is  
>(what else) a raised cosine:  
>....

At least some concrete math, Thanks Alan !

Pawel

## 21. Matlab and DigComT

From: Frode Weierud <frode@dxcern.cern.ch>  
Date: Wed, 6 Dec 1995 10:24:20 +0100 (MET)  
To: HFSIG TAPR <hfsig@tapr.org>  
Message-ID: <Pine.Ult.3.91.951206085803.254A-100000@dxcern.cern.ch>  
Rendered body type: text/plain

Hi all,

Some of you might be better off financially than me or have a possibility to acquire a tool like DigComT through your place of work. I should like to state that I have no interest in DigComT or Native Intelligence and that this posting should not be seen as an advert for DigComT, but purely information for fellow hams. You might also like to visit the DigComT WWW Home Page at <http://Ukulele.ee.cornell.edu/DCT/>

Here is a copy of my communication with Native Intelligence:

>From linda.heegard@cornell.eduWed Dec 6 08:56:02 1995  
>Date: Tue, 5 Dec 1995 23:18:35 -0500  
>From: Linda-Marie Heegard <linda.heegard@cornell.edu>  
>To: Frode Weierud <frode@dxcern.cern.ch>  
>Subject: Re: Info about DigComT

>Dear Chris,

>

>I am a radio amateur with a strong interest in the development of HF  
>digital modems using DSP techniques. I have recently bought the student  
>version of Matlab to allow me to better simulate some of these modems  
>before trying to implement them on a DSP platform.

>

>I have just heard about DigComT and I have had a quick look at what it can  
>do. It appears to fulfil all my best dreams about a simulation and  
>analysis packages, but can I run DigComT on the Matlab student version?

>As all my research is done as a radio ham and on a private, very tight  
>budget I am wondering if you have a student version of DigComT as well?

>

>Best regards,

>Frode, LA2RL

>---

>Frode Weierud Phone : 41 22 7674794  
>CERN, SL Fax : 41 22 7679185  
>CH-1211 Geneva 23, Switzerland E-mail : frode@dxcern.cern.ch

-----  
Frode,

We are not sure at this time if DigComT will run on the MatLab student version but we believe it will. If you are seriously interested in purchasing the product we will verify that it works beforehand. In any case here is some additional information that might be helpful.

The standard educational discount for a single user license is 50% off the standard price. If you intend to use the toolbox for many students in a course, a group discount is possible. I'm also willing to listen to any interesting proposals.

Let me know if I can answer any other questions for you.

-Linda-Marie

-----  
DigComT  
The Digital Communications Toolbox

Thanks for your interest in DigComT. If you want to purchase a copy, or would like more detailed information (e.g., a table of contents or a manual), please contact me directly. I have enclosed the basic price structure for the product (note: there is an educational discount available) as well as some information about it's functions and my biography. I have also attached a copy of a typical software agreement.

Please feel free to ask any questions you might have about

DigComT. I look forward to hearing from you.  
Sincerely,

Chris Heegard/Linda-Marie Heegard

-----

Price -

DigComT is priced at \$1,995 per single user copy (unix), \$895 (windows or mac). Owners are required to sign a software agreement to use this product.

-----

Introduction-

DigComT provides a variety of useful tools for the analysis, design and development of modern digital communications systems, using the commercially available program Matlab (required to use with DigComT). The system is designed to work on all the major platforms for which Matlab is available (most notably, Unix, Macintosh and Windows). The toolbox consists of a large number of Matlab ".m" files (source files of Matlab code) as well as a number of ".mex" files (binary files of "C" code that is callable in Matlab). The former files allow for the analysis of a variety of digital communications systems, while the latter allow for efficient simulation capabilities.

DigComT consists of the following packages -

\* Analysis - analytical tools

Consists of a number of analytical tools for understanding digital communications systems. Methods of estimating channels capacity and error rate performance are typical tools found here.

\* ChanlSim - channel simulation

Provides a method for modeling the noise and interference of digital communication transmission. Includes tools for modeling analog transmission systems, colored noise and channel impairments such as filtering, echos and multipath.

\* Compress - compression tools

Tools to study common analog source coding techniques (APCM, ADPCM) and other compression tools, such as Huffman coding, are provided.

\* DataFnc - data functions

Functions to manipulate and organize data. Bitwise logical functions (e.g., and, or, xor, logical\_shift) as well as buffering and mapping functions are found here.

\* DCTdemos - demonstration programs

Examples of the use of the toolbox. Useful for learning how the toolbox can be used.

\* DCTUtils - utilities

A library of common utility functions common to DigComT.

\* Discrete - discrete math function

Discrete numerical functions such as combinatorial functions (e.g., binomial and multinomial functions) and number theoretic quantities are presented.

\* FEC - forward error control

Forward error-correction tools that include: Trellis coding (convolutional codes, QAM trellis codes and Viterbi decoding), Block coding (Reed-Solomon and BCH codes) and interleaving simulation.

\* FFLib - finite fields library

A library of tools for finite field computations (useful for studying algebraic coding techniques).

\* FileFnc - file functions

A library of common file manipulation functions for use in DigComT.

\* Modem - modem simulation functions

Simulation tools for modeling modulation and demodulation processes. A variety of modulations are supported (PAM, QAM, PSK, ...). Tools for developing receiver structures involving AGC, carrier recovery, symbol timing recovery and adaptive equalization are provided.

\* Numeric - Numerical functions

Numerical functions used in the study and simulation of digital communications systems. Examples include quantizers, mean-squared estimators (LMS and RLS), and information theoretic functions (entropy and mutual information).

\* PlotFnc - plotting functions

A library of plotting functions common to DigComT. Examples include functions to plot signal constellations and eye patterns.

\* RandSig - random signal functions

Deals with methods of generating and manipulating a variety of random signals commonly required for digital communications.

\* Signals - deterministic signal functions

Deals with methods of generating and manipulating a variety of deterministic signals commonly required for digital communications.

-----

About the Author-

Chris Heegard was born in Pasadena, California, on October 4, 1953. He received the BS and the MS degrees in electrical and computer engineering from the University of Massachusetts, Amherst, MA, in 1975 and 1976, respectively, and the Ph.D. degree in electrical engineering from Stanford University, Stanford, CA, in 1981.

From 1976-1978, he was an R&D Engineer at Linkabit Corp., San Diego, CA, where he worked on the development of a packet switched satellite modem and several sequential decoders for the decoding of convolutional codes. In 1981, he joined the faculty of the School of Electrical Engineering, Cornell University, Ithaca, NY, as an Assistant Professor; he was appointed to Associate Professor, with tenure, in 1987.

At Cornell, Professor Heegard teaches courses in digital communications, error control codes, information theory, detection and estimation theory, digital systems and audio engineering. His current research interests include: information, coding and communication theory, algorithms for digital communications, coding for computer memory systems, signal processing and error-control in optical and magnetic recording systems, audio and video signal compression, algebraic geometric coding theory and symbolic and numerical computer methods.

Dr. Heegard is an active member of the consulting community. He has worked on problems of digital HDTV and cable TV transmission, DSP and hardware based trellis coded modems, modulation and error-control for optical LANs, and modulation and coding for recording systems.

In 1984, Dr. Heegard received the Presidential Young Investigator Award from the National Science Foundation and the IBM Faculty Development Award. He has ongoing research support from the NSF as well as ARO and NSA. Chris has been involved in the organization of several IEEE workshops and symposia. In 1986, he was elected to the Board of Governors of the Information Theory Society of the IEEE and reelected in 1989. In 1994 he was IT President; he is currently Past-President of the group. He is a fellow of IEEE, and a member of AES and Eta Kappa Nu.

-----

Trademark/Copyright notice -

No part of this program or manual can be copied, used or altered without the express written consent of the author. DigComT is not to be used without license from Native Intelligence. DigComT, The Digital Communications Toolbox and Native Intelligence are trademarks of Native Intelligence. Matlab is a registered trademark of MathWorks Inc.

-----

A Typical Software Agreement -

DigComT  
The Digital Communications Toolbox

In using The Digital Communications Toolbox , we \_\_\_\_\_ (XX)  
agree to the following terms:

\* This is a single user license for this software, running only under a licensed version of Matlab from the MathWorks Inc., (24 Prime Park Way, Natick, MA 01760, (518) 653-1415)

\* Native Intelligence (NI) supplies all the ".m" files, documentation files and ".mex" files. The ".mex" files are provided in binary form only. It is understood that these file are copyrighted and are not to be distributed, in original or altered form, without the express written permission of NI.

\* A version of the documentation is supplied. It is understood that the documentation is copyrighted and is not to be distributed, in original or altered form, without the written permission of NI.

\* NI provides a number of test programs to demonstrate how these routines are to be used within Matlab.





This is too expensive for us, too...

I use "ptolemy" under AIX and linux. This is free software and really fast compared to Matlab based systems (which is basically an interpreter).

Have a look at <http://ptolemy.eecs.berkeley.edu/>

I have made channel simulators for the HF channel (Watterson model) running under ptolemy. Currently I'm analysing different OFDM systems (frequency and time differential modulation).

BTW my realtime HF channel simulator is running! One of our students did it as his final project. The simulator is running on a DSPCOM (40MHz TMS320C26). It should run on the c26-EVM board with some small changes. I hope I can post the sources (after some small changes) to the ftp-site.

73' Alexander

```

| Alexander F. Kurpiers |
Institut f. Uebertragungstechnik	Voice: +49-6151-162369
u. Elektroakustik	Fax : +49-6151-165545
Merckstrasse 25	EMail: a.kurpiers@uet.th-darmstadt.de
D-64283 Darmstadt (Germany)	Hamradio: dl8aa@db0zdf.#rpl.deu.eu

```

## 24. Re: [HFSIG:749] M-ary FSK interference

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 07 Dec 1995 14:25:33 -0800  
To: hfsig@tapr.org, hfsig@tapr.org  
Message-ID: <199512071842.KAA20288@qualcomm.com>  
Rendered body type: text/plain

At 10:45 AM 12/6/95 -0600, Richard Lay wrote:

>Hello.

>Does anyone know a reference discussing the performance of M-ary FSK with  
>various types of interference? I'm looking for theoretically, simulated, or  
>experimentally derived curves (or equations) showing the probability of  
>symbol error given a signal to interference ratio. I've found such a  
>reference when the interference is gaussian. What if the interference is CW  
>or linear FM? What if it's something else, other than gaussian?

Yes. Several of the textbooks on modulation and coding theory discuss the theory and performance of m-ary orthogonal modulation, of which M-ary FSK is an example. (All m-ary orthogonal schemes perform exactly the same as long as the implementation losses are the same).

A few good references are Digital Communications by Proakis (the 3rd edition just came out) and my personal favorite, Digital Modulation and Coding by Wilson (also brand new).

I will assume noncoherent demodulation, which is the usual reason for using M-ary orthogonal modulation. The demodulator correlates the incoming signal against reference copies of each of the orthogonal basis functions (e.g., carrier frequencies for M-ary FSK), accumulates the resulting energies over the symbol time into M bins, and takes the bin with the largest result as the output symbol. The probability of a correct decision is therefore the probability that the correct bin energy exceeds all the incorrect bin energies.

For M-ary FSK, the demodulator can just be the FFT with each bin containing the sum of the squares of the sine and cosine components.

In gaussian noise, the bin energies will be Rayleigh distributed for all the "wrong" channels and Ricean for the correct bin. (The Ricean distribution resembles a Rayleigh distribution when the S/N ratio is low, and it becomes a nonzero mean Gaussian when the S/N is high.) You analyze the performance of the demodulator in noise by basically looking at the areas under the distribution tails. You can do this by analytical means, but it's pretty hairy (even though the result is simple) because of the nonlinearity in the comparisons. And it's already been done in the books I mentioned.

For nongaussian noise, a Monte Carlo simulation is probably the most practical. In CW interference, you'll get the right answer only if the energy in the right bin exceeds the CW energy in the wrong bin (assuming the CW is much stronger than

the noise, of course).

To improve performance against interference, you could apply a forward error correcting code plus a whitening filter. That is, when the desired transmitter is off the air, you run the demod for a while and look at the energies in each bin. In white noise (or QRM) they will be approximately equal. If the noise isn't white (e.g., with strong CW QRM) one or more bins will greatly exceed the others. "Whiten" the signal before demodulation by adjusting the gain of each bin so they're again roughly equal.

Since this causes the desired signal to be notched out when it happens to land on the CW QRM, you'll occasionally get errors. That's what the FEC is for. As long as you scramble the data such that each bin (tone) is equally likely, then your probability of error due to QRM will be something like  $1/N$  where  $N$  is the number of bins (tones), and if this is within your error correction capability you'll get error free data despite the QRM. Assuming of course that the QRM isn't so strong that intermodulation distortion or FFT "leakage" blots out multiple bins.

Again, Monte Carlo simulation is probably your best bet for getting real numbers when the situation is complex, e.g., if the interference is changing frequency or amplitude.

The nonlinearity in the bin comparison is what makes it impossible to improve noncoherent demodulation performance indefinitely by lowering the FEC code rate, unlike coherently demodulated PSK.

At some point the increased loss due to the demodulator nonlinearity more than swamps the extra coding gain from the lower rate FEC. The optimum code rate depends on the value of  $M$  and whether the channel is AWGN or fading. For typical values of  $M$  (16 or 32) on AWGN, there's a broad optimum around rate  $1/2$ . For a Rayleigh fading channel the optimum is narrower and depends more strongly on  $M$ ; for  $M=32$  the optimum is about rate  $1/10$ . Wilson's book has an excellent set of graphs that show this.

Phil

---

## 25. Re: [HFSIG:750] Re: Matlab and DigComT

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 08 Dec 1995 15:30:35 -0800  
To: hfsig@tapr.org  
Message-ID: <9512082326.AA24952@ucs.orst.edu>  
Rendered body type: text/plain

Hello Alexander,

I have a brief break (a day or two) - so time to catch up.

>  
>>Some of you might be better off financially than me or have a possibility  
>>to acquire a tool like DigComT through your place of work. I should like  
>  
>This is to expensive for us, too...  
>  
>I use "ptolemy" under AIX and linux. This is free software and really  
>fast compared to Matlab based systems (which is basically an interpreter).  
>  
>Have a look at <http://ptolemy.eecs.berkeley.edu/>

How big a Linux box do you need (memory and disk space) to run Ptolemy ?  
There also is a PD Matlab -> C++ translator available that I have used in  
the past. It has a shell that allows you to use the regular Matlab  
interpreter, when you get your code working, translate it to C++ and it is  
then an executable instead of interpreted.

>  
>  
>I have made channel simulators for the HF channel (Watterson modell)  
>running under ptolemy. Currently I'm analysing different OFDM systems  
>(frequency and time differential modulation).  
>  
>  
>BTW my realtime HF channel simulator is running! One of our  
>students did it as his final project. The simulator is running on a DSPCOM  
>(40MHz TMS320c26). It should run on the c26-EVM board with some  
>small changes. I hope I can post the sources (after some small  
>changes) to the ftp-site.

>

This is fantastic news! Congratulations - we are looking forward to see how it works.

I assume this is the Rayleigh fading model, have you given any thought yet to the Markovian impulse noise model that follows the Rayleigh fading?

>

>73' Alexander

>

>

```
>*-----+-----*
>| Alexander F. Kurpiers |
>| Institut f. Uebertragungstechnik | Voice: +49-6151-162369 |
>| u. Elektroakustik | Fax : +49-6151-165545 |
>| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de |
>| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu |
>*-----+-----*
```

>

>

>

>

73's,

--Johan, KC7WW

---

## 26. BPSK or DBPSK ?

From: "Pawel Jalocho" <jalocho@home.ifj.edu.pl>  
Date: Fri, 15 Dec 95 12:54:19 MET  
To: dsp4@nic.funet.fi, hfsig@tapr.org  
Message-ID: <1336.jalocho@home.ifj.edu.pl>  
Rendered body type: text/plain

Which modulation scheme is better: straight BPSK or differential BPSK ?

>From some discussion in the past the conclusion was that a differential phase keying scheme is always worse by 3 dB because you rely on two successive symbols and both of them are being affected by noise thus the difference contains twice as much noise.

However I often see sentences saying that differential BPSK is better because the the difference between the two states is greater from the distance of a single state from base (zero).

Or does it depend on the demodulator type ?

I naively make a dot product of two successive I/Q vector  
- is there a better way ?

A related question which is bothering me: is there any S/N advantage by doing the Manchester encoding before applying the data to the BPSK modulator like they do in the 400 bps telemetry ?

Surely the signal's spectrum becomes twice as wide, does this gain any S/N ?

The 1200 bps BPSK modem for satellites is almost ready.

Pauli Jeskanen has done lot of tests and he adjusted some paramaters.

Some bugs/features have been found and corrected on the way.

Yesterday the modem seemed to transmit and receive right.

Some more cosmetic changes and the modem is ready.

The modem seems to perform at least as good as the JAS-1 modem (but I have no idea how good the JAS-1 is).

The satellite modem for for the DSPCARD4/EVM56K can be used for other bands/modes for example for the HF, and it can be scaled down to 600 or 300 bps. One good test I was thinking about is to compare how the 300 bps HF packet performs with BPSK (my modem) and FSK (an ordinary TNC).

Pawel

---

## 27. Book recommendation sought

From: gc@fox.cen.com (Gary Chatters)  
Date: Fri, 15 Dec 95 10:01:15 EST  
To: hfsig@tapr.org  
Message-ID: <9512151501.AA04128@cen.com>  
Rendered body type: text/plain

Hello everyone,

I have been reading this list for a few weeks now. It is one of the more interesting maillists available through Internet.

I am hoping that someone can recommend one or two good books on digital data communications that I could read to get better understanding of the topics discussed here. There are probably many books on the subject, but only a few would likely be well written and have a good coverage of all the basic information. The book should cover things like generation and detection of various modulation schemes (FSK, PSK, QPSK,...). Error correction coding would be interesting, too, but that might be another book.

About a week ago in response to a query about M-ary FSK  
Phil Karn wrote:

>  
>A few good references are Digital Communications by Proakis (the 3rd edition  
>just came out) and my personal favorite, Digital Modulation and Coding by  
>Wilson (also brand new).  
>

Perhaps this is the answer to my question right there. Or are these books rather specialized? Any other recommendations?

Note: Basic math, at least through differential equations, complex variables, etc. is not a problem. (I haven't got wavelets figured out yet though :-) ).

BTW, do you people do any regular on the air HF activity?

TNX ES 73,

Gary

---

## 28. Re: [HFSIG:754] Book recommendation sought

From: Greg Jones <wd5ivd@tapr.org>  
Date: Fri, 15 Dec 1995 10:35:52 -0600 (CST)  
To: hfsig@tapr.org  
Message-ID: <199512151635.KAA03561@sys1.tapr.org>  
Rendered body type: text/plain

TAPR will be publishing a book by Tom McDermott entitled "Digital Radio Communications: Design and Theory." There was a full table of contents printed in the Summer issues of the TAPR PSR I believe.

It should fit exactly what you are looking for from you comments.

Looks like maybe 300 pages, since I am over 150 and only at Chapter 7 of 15.

Cheers - Greg

>  
> Hello everyone,  
>  
> I have been reading this list for a few weeks now. It is one of the  
> more interesting maillists available through Internet.  
>  
> I am hoping that someone can recommend one or two good books on  
> digital data communications that I could read to get better understanding  
> of the topics discussed here. There are probably many books on the  
> subject, but only a few would likely be well written and have a  
> good coverage of all the basic information. The book should cover  
> things like generation and detection of various modulation schemes  
> (FSK, PSK, QPSK,...). Error correction coding would be interesting, too,  
> but that might be another book.  
>  
> About a week ago in response to a query about M-ary FSK  
> Phil Karn wrote:  
> >  
> >A few good references are Digital Communications by Proakis (the 3rd edition  
> >just came out) and my personal favorite, Digital Modulation and Coding by  
> >Wilson (also brand new).  
> >  
> >  
> Perhaps this is the answer to my question right there. Or are these  
> books rather specialized? Any other recommendations?  
>

> Note: Basic math, at least through differential equations, complex  
> variables, etc. is not a problem. (I haven't got wavelets figured  
> out yet though :-)) ).  
>  
> BTW, do you people do any regular on the air HF activity?  
>  
> TNX ES 73,  
>  
> Gary  
>  
>

---

## 29. RE: [HFSIG:753] BPSK or DBPSK ?

From: rick@itron-ca.com  
Date: Fri, 15 Dec 95 09:03:56 PST  
To: hfsig@tapr.org  
Message-ID: <Chameleon.4.01.951215091900.rick@rick.itron-ca.com>  
Rendered body type: text/plain

Pawel,

1. DBPSK and PSK differ in that errors occur in PAIRS with DBPSK but can occur singly in BPSK. The differential business means that making a mistake in estimating the phase of one bit means the next bit will screw up too since the first bit is the reference for the second bit. Because the errors occur in pairs, however, only implies that the BER for a given SNR is double, this does not imply that the SNR needs to be 3 dB larger in order to get the same error rate! At a BER of .001 the difference amounts to only about a 1 dB loss in SNR and at .00001 the difference is amazingly negligible.
2. DBPSK can be demodulated coherently or non coherently. Non coherent methods require a delay line that is as long as a bit time. The difference between the transmitted carrier and the receiver LO, however, must be stable over a bit time, in other words, the drift of phase of the carrier over the bit time must be small. 10 degrees or so should do it. This gets harder to do as the data rate gets lower and lower which is one of the reasons why DBPSK is often used with high speed microwave links.
3. A popular user of DBPSK is GPS which uses it for the 50 BPS data downlink (if my memory serves me correctly).
4. Manchestered data offers no improvement in SNR, however, it makes bit sync a lot easier (at the expense of 2x the bandwidth as you have noticed). In so far as sync errors degrade detection then Manchesterizing the data is good if it will fit into the bandwidth allocation.

Cheers!

Rick W6NZK

-----  
Rick Booth  
'95 900SS SP  
E-mail: rick@itron-ca.com  
-----

---

## 30. Re: [HFSIG:754] Book recommendation sought

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 15 Dec 1995 09:48:27 -0800  
To: hfsig@tapr.org  
Message-ID: <9512151743.AA18156@ucs.orst.edu>  
Rendered body type: text/plain

Hi Gary,

Welcome to HFSIG. Glad you found us.

>Hello everyone,  
>  
>I have been reading this list for a few weeks now. It is one of the  
>more interesting maillists available through Internet.  
>  
>I am hoping that someone can recommend one or two good books on  
>digital data communications that I could read to get better understanding  
>of the topics discussed here. There are probably many books on the  
>subject, but only a few would likely be well written and have a  
>good coverage of all the basic information. The book should cover  
>things like generation and detection of various modulation schemes  
>(FSK, PSK, QPSK,...). Error correction coding would be interesting, too,  
>but that might be another book.  
>  
>About a week ago in response to a query about M-ary FSK  
>Phil Karn wrote:  
>>  
>>A few good references are Digital Communications by Proakis (the 3rd edition  
>>just came out) and my personal favorite, Digital Modulation and Coding by  
>>Wilson (also brand new).

>>  
>  
>Perhaps this is the answer to my question right there. Or are these  
>books rather specialized? Any other recommendations?

I suppose you will get a number of good suggestions from the group. The following three books have been quite useful to me - I picked them up at our local used bookstore:

- 1) Digital Communications Fundamentals and Applications. Bernard Sklar. Prentice Hall. ISBN 0-13-211939-0 025. Carries a common thread throughout the book - Highly recommended.
- 2) Telecommunication Transmission Handbook. Third Edition. Roger L. Freeman. Wiley. ISBN 0-471-51816-6. General topics includes theory and a lot of good stuff on the HF channel, different types of modems, coding etc. - much lighter reading.
- 3) Telecommunication Systems Engineering. Lindsey and Simon. Dover. ISBN 0-486-66838-X. A compendium of the works of these authors. Covers a lot of theory on the mechanics of PSK detection, clock recovery, Costas loops - more than you will ever need to know. No DSP, but heavy reading.

These, however, does not really specialize in DSP techniques, for that topic there are much more specialized literature such as Messerschmidt's book on modem design for example that I just have to mention. It does not take much to realize that this field have become a very involved mish-mash of different disciplines, mathematics being the cornerstone. It also is quite obvious that DSP techniques holds tremendous possibilities for the future.

>  
>Note: Basic math, at least through differential equations, complex  
>variables, etc. is not a problem. (I haven't got wavelets figured  
>out yet though :-) ).  
>  
>BTW, do you people do any regular on the air HF activity?

The Fin's have been doing some innovative testing of OFDM, but things are only just beginning. It's a good time to get your feet wet. I suspect that a lot of interesting things will be happening in the near future.

Keep in touch.

--Johan Forrer, KC7WW

---

### 31. New PCMCIA DSP module

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 15 Dec 1995 09:48:44 -0800  
To: hfsg@tapr.org  
Message-ID: <9512151743.AA18147@ucs.orst.edu>  
Rendered body type: text/plain

Hi all,

Thought that I would pass on a tidbit of exciting news. I have no relation with the following company nor do I get any special treatment, just passing on the news.

CAC now manufactures and sells a type 3 PCMCIA DSP module called the Bulletdsp. It is based on the TI320C3x floating point DSP and contains a 16-bit stereo CODEC, SRAM, DRAM, and FLASH memory. The base price of the Bullet is \$895 and comes with 256KB SRAM, 0 DRAM, 128KB flash memory, the 50MHz C32 and the stereo codec. Its part number is Bulletdsp-S2D0F1.

Replacing the -S2 with a -S1 (a \$700 option) increases the SRAM to 1MB.  
Replacing the -D0 with a -D2 (a \$300 option) increases the DRAM to 2MB.  
Replacing the -D0 with a -D4 (a \$600 option) increases the DRAM to 4MB.  
Replacing the -F1 with a -F5 (a \$ 75 option) increases the flash to 512KB.  
The 60MHz version and the PCMCIA type II are not yet available.

The user's manual, which includes the MS-DOS and Windows 3.1 drivers, comes with the Bulletdsp. The software also contains several diagnostic and demo programs, such as audio record/playback and FFT. Socket services software are not provided (this should have been included with your notebook computer.) Refer to the "Programming Interface" for more information about CODEC software that is included with the Bulletdsp.

Any TI C3x C-compiler or assembler works with our board. We offer the TI C-compiler for \$1500, part number TIC32-CDS, and the assembler for \$500, part number TIC32-DS.

For further details, please contact, CAC - Jim Bridges  
(71342.1717@compuserve.com).

Wouldn't this module in conjunction with a laptop would make a powerful portable DSP platform?

73's

--Johan Forrer, KC7WW

---

## 32. Re: Book recommendation sought

From: rick@itron-ca.com  
Date: Fri, 15 Dec 95 13:30:42 PST  
To: hfsig@tapr.org  
Message-ID: <Chameleon.4.01.951215133236.rick@rick.itron-ca.com>  
Rendered body type: text/plain

Another book that includes both DSP and communications:

Digital Signal Processing Communication Algorithms, Frerking, ISBN 0-442-01616-6

Rick  
W6NZK

-----  
Rick Booth  
'95 900SS SP  
E-mail: rick@itron-ca.com  
-----

---

## 33. Re: [HFSIG:741] Re: BPSK modem for 1200 bps

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Mon, 18 Dec 1995 07:58:46 +0200  
To: hfsig@tapr.org  
Message-ID: <199512180558.HAA02757@f15.pix.za>  
Rendered body type: text/plain

Hi Pawel

Yes I would of course be interested in FEC. I want to learn more about the topic in any way. Are you going to use block codes or convolutional codes ? If you use block codes then one can perhaps do it on the PC side and spare the 56002 the effort ? :-)

You are right about the formula for raised cosine. The formula for root raised cosine is:

$$h(t) = (4a/\pi \sqrt{T}) * ((\cos(1+a)\pi t/T) + (T \sin(1-a)\pi t/T) / (4at)) / (1 - (4at/T)^2);$$

$a=0$  is the ideal pulse shape.

Hell what a long formula ! Look on p229 of Lee&Messerschmidt of 'Digital Communication'. I am sure there must be a simpler equation ?

Regards, Danie

>>We will have to ask James Miller G3RUH. My documentation says just raised  
>>cosine but your are of course right. The root is normally taken.  
>  
>So what is mathematically the root raised cosine ?  
>Is it  $\sin(x)/x$  ?  
>  
>The other thing is that I would like to know which filter to use  
>on the \_receiver\_ side. Suppose the transmitter is raised cosine:  
> $0.5 + 0.5 \cos(x)$  for  $x = -\pi \dots \pi$  what should be the receiver's shape ?  
>I suspect you need  $\sin(x)/x \dots$  but this one is infinitely long !  
>  
>Over the weekend I scaled the BPSK modem the other way, down  
>to 30 bps. Then I made up a simple terminal interface:  
>- pressing Ctrl-T keys the transmitter  
>- pressing Ctrl-R releases the PTT  
>- any other character gets transmitted on the carrier as a 7-bit ASCII  
> in asynchronous-like format.  
>- Any received character is printer on the screen

>  
>The signal's bandwidth is some 30 Hz at -6dB and 60 Hz at -50 dB.  
>The specific operating feature is that the receiver sweeps its carrier  
>within given margins to find the lock and this takes time...  
>for +/- 20 Hz sweep the lock may take 5-15 seconds.  
>The transmitting station needs to send that much of idle/dummy  
>information.  
>  
>If there is some interest I could go on and add some FEC...  
>  
>Pawel  
>  
>

---

### 34. Re: [HFSIG:739] Re: Negative Frequencies

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Mon, 18 Dec 1995 07:58:43 +0200  
To: hfsig@tapr.org  
Message-ID: <199512180558.HAA02755@f15.pix.za>  
Rendered body type: text/plain

Hi Phil.

Thanks for the answer. I think we are now on the same wavelength :-) My question was regarding a simple-to-calculate routine. Too compute arctan in a DSP chip is time consuming or not ?

Any way I have found an algorithm that works but it is very sensitive for amplitude variation. If you are interested I will send you the Matlab code. It boils down to something like this:

```
y(t)=r1+jIm1
y(t+1)=r2+jIm2
```

the sign of the following determine the direction of the phasor:  
 $(Im2*r1 - Im1*r2)$ .

One can thus say whether the freq is neg. or pos.

I implemented a FSK demod using it and I could recover the data again but it was very sensitive for input signal strength.

Regards , 73 de danie zs6awk

PS Please note - I am just learning. The things am trying to do have been done countless times by others :-)

>At 11:24 PM 12/1/95 -0600, you wrote:  
>  
>>y(t)=re1+jim1  
>>y(t-1)=re2+jim2  
>>  
>>Now which way is the phasor turning ? Clockwise or anticlockwise ?  
>  
>Seems pretty straightforward to me. Assuming no aliasing (an important assumption),  
>all you have to do is convert the two samples to polar notation, computing just  
>the angle part since that's all we're interested in here:  
>  
>phi(t) = atan2(im1,re1)  
>phi(t-1) = atan2(im2,re2)  
>  
>where atan2(y,x) = arctangent(y/x) with automatic quadrant placement. (Many  
math  
>libraries since Fortrash days have this function built in.)  
>  
>Now you just subtract the two angles mod  $2\pi$ . If one is, say,  $\pi/2$  larger than  
>you can consider the vector to have rotated in that direction. Of course,  
it's also  
>possible that it rotated  $3\pi/2$  (270 deg) in the opposite direction, which is  
>why it's important to assume no aliasing.  
>  
>Phil  
>  
>

---

### 35. Re: [HFSIG:743] Motorola DSP?



From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Mon, 18 Dec 1995 07:59:22 +0200  
To: hfsig@tapr.org  
Message-ID: <199512180559.HAA02786@f15.pix.za>  
Rendered body type: text/plain

Hi

Try this email. This person is the Motorola person for Africa but she will be able to tell you who to contact:

Shuba Iyer  
shuba@cadsun.corp.mot.com

Regards, Danie ZS6AWK

> I'm looking for some General/Technical information about the  
> Motorola DSP 56002EVM. I've searched Motorola's Home Page and found  
> nothing. Thanks!  
>  
> --73's  
> V/R DMP  
>  
>

---

### 36. (no subject)

From: "Allan E. Matlick" <matlick@gate.net>  
Date: Mon, 18 Dec 1995 13:45:00 -0500  
To: hfsig@tapr.org  
Message-ID: <199512181845.NAA17674@osceola.gate.net>  
Rendered body type: text/plain

cancel

-----  
Allan E. Matlick  
4040 Crockers Lake Blvd. 1722  
Sarasota, FL 34238-5543  
Tel: 941-923-7008  
Fax: 941-927-8178  
Internet: matlick@ gate.net  
MSN:Allan\_Matlick@msn.com  
CompuServe: 71630,1561  
-----

---

### 37. DSP56002EVM

From: David Wilson Borden <dwborden@ns1.panenet.com>  
Date: Mon, 18 Dec 1995 17:10:41 -0500  
To: hfsig@tapr.org  
Message-ID: <30D5E6E1.6AE2@panenet.com>  
Rendered body type: text/plain

The Product Brief for the 24-Bit DSP56002 Evaluation Module is available free from Motorola. If you get to the Motorola DSP Home Page, you can follow to the literature ordering form and order the following:  
DSP56002EVMP/D Rev.1  
It describes the EVM in enough detail to convince you that you need it.

> I'm looking for some General/Technical information about the  
> Motorola DSP 56002EVM. I've searched Motorola's Home Page and found  
> nothing. Thanks!  
>  
> --73's  
> V/R DMP  
>  
>

---

### 38. Re: [HFSIG:761] Re: Negative Frequencies

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)  
Date: Tue, 19 Dec 95 10:16:15 CST  
To: hfsig@tapr.org  
Message-ID: <9512191616.AA02275@eagle.aud.alcatel.com>  
Rendered body type: text/plain

> Thanks for the answer. I think we are now on the same wavelength :-) My  
> question was regarding a simple-to-calculate routine. Too compute arctan in  
> a DSP chip is time consuming or not ?  
>

```

> Any way I have found an algorithm that works but it is very sensitive for
> amplitude variation. If you are interested I will send you the Matlab code.
> It boils down to something like this:
>
> y(t)=r1+jIm1
> y(t+1)=r2+jIm2
>
> the sign of the following determine the direction of the phasor:
> (Im2*r1-Im1*r2).
>
> One can thus say whether the freq is neg. or pos.
>

```

Danie: I think that your formula may be correct - at least it seems to work on a few examples. In general, the cross-product of two vectors should indicate the direction of rotation about the origin from one vector to the other. Thus, if we let  $y(t) = V1$  (vector one) and  $y(t+1) = V2$ , then

$V3 = V1 \times V2$  where  $V3$  is the cross-product of the two vectors.

Then the direction that  $V3$  points should indicate the direction of rotation. Since  $V3$  should be normal to the (re,im) plane, it should be possible to take the sign of  $V3$  as indicative of the direction of rotation. Now, it's been 24 years since I last did any of this stuff (in physics at UC Berkeley) so I'm a bit rusty. Also, I don't have my physics book handy. Your formula may indeed be the cross-product in the z-axis. Maybe one of the readers will have a better memory than I and can perform the computation. If this is correct, then it is probably efficient to compute.

```

> I implemented a FSK demod using it and I could recover the data again but it
> was very sensitive for input signal strength.
>
> Regards , 73 de danie zs6awk
>
> PS Please note - I am just learning. The things am trying to do have been
> done countless times by others :-)
```

We are all (still) learning!

- Tom, N5EG

```

-----+-----
Tom McDermott | "All opinions expressed
Alcatel Network Systems, Inc. | are my own, and do not
mcdermot@aud.alcatel.com | represent those of Alcatel
[ICC'96 Technical Program Secretary] | Network Systems, Inc."
[June 23-27, 1996, Dallas, Tx.] |
-----+-----

```

### 39. Re: [HFSIG:753] BPSK or DBPSK ?

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 19 Dec 1995 18:14:37 -0800 (PST)  
To: hfsig@tapr.org  
Message-ID: <199512200214.SAA16678@servo.qualcomm.com>  
Rendered body type: text/plain

>Which modulation scheme is better: straight BPSK or differential BPSK ?

Coherently demodulated BPSK is always better than differentially decoded PSK. Assuming, that is, you have a perfect carrier phase reference at the receiver. The whole reason for using DPSK is to avoid the need for such a reference, either because it's too hard to build one or because the channel phase characteristics are so unstable (e.g., Rayleigh fading) that carrier phase extraction is not practical.

The penalty for DPSK vs BPSK depends on the S/N. At high S/N ratios the two are nearly indistinguishable because there is little noise on the "reference" signal, so DPSK performs much like BPSK with a good reference.

At low S/N ratios the curves diverge because of the noisy "reference". (Of course, a practical BPSK system will also diverge from ideal because it too will have an imperfect phase reference).

>A related question which is bothering me: is there any S/N advantage by doing  
>the Manchester encoding before applying the data to the BPSK modulator  
>like they do in the 400 bps telemetry ?  
>Surely the signal's spectrum becomes twice as wide, does this gain any S/N ?

Offhand I'd say this has no effect on S/N as long as matched filter detection is used in both cases. The two Manchester signal states are still antipodal.

It's easy to see this with a little thought. You can produce Manchester encoding of a baseband modulated BPSK signal by simply multiplying it again by a square wave at the data rate. And you can undo this at the receiver by multiplying again by the same square wave signal. Assuming perfect phasing of the square wave signal, the demodulator wouldn't even know it was there. The performance is exactly the same in either case.

It's like direct sequence PN spreading, which has no effect on the S/N requirements as long as the channel is linear and the demodulators are all ideal matched filters. In fact, Manchester \*IS\* direct sequence spreading, only the "spreading sequence" is just two chips long, repeats on every data bit, and consists of the pattern 01010101....

Phil

---

#### 40. Re: [HFSIG:754] Book recommendation sought

From: Phil Karn <karn@qualcomm.com>  
Date: Tue, 19 Dec 1995 18:28:40 -0800 (PST)  
To: hfsig@tapr.org  
Message-ID: <199512200228.SAA16689@servo>  
Rendered body type: text/plain

>>

>>A few good references are Digital Communications by Proakis (the 3rd edition  
>>just came out) and my personal favorite, Digital Modulation and Coding by  
>>Wilson (also brand new).

>Perhaps this is the answer to my question right there. Or are these  
>books rather specialized? Any other recommendations?

These books are both written for a senior or graduate level digital communications theory course. They're at about the same level, though I find Wilson somewhat easier to read.

Phil

---

#### 41. Re: Negative Frequencies

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)  
Date: Wed, 20 Dec 95 08:15:42 CST  
To: hfsig@tapr.org  
Message-ID: <9512201415.AA02556@eagle.aud.alcatel.com>  
Rendered body type: text/plain

Danie: as a follow-up to yesterday's message I broke out the freshman physics book, and here's the derivation of the formula:

Assume two vectors, A and B. Define the vector A (in 3-dimensions) as follows:

$$A = a_x X + a_y Y + a_z Z$$

where X, Y, and Z are the three directions on the coordinate system.  $a_x$  is the component of vector A in the X direction,  $a_y$  is the component in the Y direction, and  $a_z$  is the component in the Z direction. We'll assign X to be the real axis, Y to be the imaginary axis, and Z to be normal to X and Y (Z points out of the page formed by the real, imaginary plane. In the right-handed coordinate system, X goes from center to the right, Y goes from the center up the page, and Z comes out of the page towards the reader.

Assuming positive frequency, then vector A (at time t) rotates to position B (at time t+1) by making counter-clockwise rotation.

Then we form the two signals at time (t) and time (t+1) and assign each to vector A and B, respectively. Assume that ( $re1, im1$ )

is at time (t) and (re2,im2) is at time (t+1), then:

$$\begin{aligned} A &= \text{re1 } X + \text{im1 } Y + 0 Z \\ B &= \text{re2 } X + \text{im2 } Y + 0 Z \end{aligned}$$

Note that there is no component of either vector in the Z axis, so that part is zero.

Let vector  $C = A \times B$  (C equals A cross B).

$$C = c_x X + c_y Y + c_z Z$$

Then it can be shown that the three component parts of C are:

$$\begin{aligned} c_x &= a_y b_z - a_z b_y \\ c_y &= a_z b_x - a_x b_z \\ c_z &= a_x b_y - a_y b_x \end{aligned}$$

Since we know that  $a_z$  and  $b_z$  are both zero, then  $c_x$  and  $c_y$  must both be zero, leaving only the  $c_z$  term. Thus the cross-product vector is normal to the re,im plane (as I correctly remembered yesterday). When  $A \rightarrow B$  rotates counter-clockwise, the Z axis term will be positive. IE: +Z equals positive when the frequency is positive, and is negative when frequency is negative.

Translating  $a_x$ ,  $b_y$ ,  $a_y$ ,  $b_z$  to the real and imaginary parts yields:

$$c_z = \text{re1 im2} - \text{im2 re1}$$

This checks with the formula that you presented, and so your formula is correct. A couple of notes:

The magnitude of  $A \times B$  is  $AB \sin(\theta)$ . Thus when the angle between the two vectors approaches zero or approaches 180 degrees, the cross product goes to zero. Also, when the vector travels more than 180 degrees between time samples, the sign reverses, so that it appears to be negative frequency. This means that the nyquist sampling theorem must be satisfied. Thirdly, this only applies to narrowband signals, not broadband signals, whose time shape does not bear the direct relationship.

Thanks for the interesting exercise, Danie. My knowledge of the subject has been increased significantly by this discussion. Good luck with your project.

- Tom, N5EG

|                                        |                            |
|----------------------------------------|----------------------------|
| -----+-----                            |                            |
| Tom McDermott                          | "All opinions expressed    |
| Alcatel Network Systems, Inc.          | are my own, and do not     |
| mcdermot@aud.alcatel.com               | represent those of Alcatel |
| [ ICC'96 Technical Program Secretary ] | Network Systems, Inc."     |
| [ June 23-27, 1996, Dallas, Tx. ]      |                            |
| -----+-----                            |                            |

## 42. Re: BPSK or DBPSK ?

From: jra1854@tntech.edu (Jeffrey Austen)  
Date: Wed, 20 Dec 1995 15:26:46 -0600  
To: hfsig@tapr.org  
Message-ID: <v01530500acf737cab3af@[149.149.42.11]>  
Rendered body type: text/plain

>Which modulation scheme is better: straight BPSK or differential BPSK ?  
>>From some discussion in the past the conclusion was that a differential  
>phase keying scheme is always worse by 3 dB because you rely on two

There are two effects, differential signaling and differentially coherent demodulation, at work.

Differential signaling doubles the error rate. As receiver takes the difference between the previous bit and the current bit, one error in the channel results in an two errors at the output. In a typical system, at low error rates, this results in a fraction of a dB worse performance.

Differentially coherent detection also degrades the performance. For a binary system the degradation at low error rates is a little less than 1 dB. In an M-ary system the degradation approaches 3 dB for large M.

Jeff, k9ja

### 43. Re: [HFSIG:769] Re: BPSK or DBPSK ?

Gary

Phil

## Page 453

```
> ^^^^^^^^^^^^^^^^^
>
>What type of a link is expected for HF? This statement surprised me
>a bit, because I think of typical HF links as having to work at low SNR
>(or at least quite variable SNR) and thus high BER.
```

More importantly, it seems to me, is the multi-path induced phase jitter. Many times, particularly on the lower HF freqs, I have experienced links with excess SNR (above that which is needed for a low BER) behave poorly with Clover due to excessive multi-path distortion.

```
>
>I have not been on this list long enough to know what kind of
>link people expect to work with. My HF experience is limited
>to CW and RTTY. Signals on RTTY can have good SNR, but you would
>like a system to work to as low an SNR as possible.
>
>Gary
>
```

Does anyone have experience with HAL's PCI-4000/M in the Clover and FSK modes? HAL's DSP implementation of FSK filters on the PCI-4000/M seems to be lacking. In my comparisons with the PK-232, e.g., it generally performs poorly. The PK-232 is not a particularly good reference model! My 35 year old vacuum tube TU performs better than the PK-232 on 45 Baud RATT.

Clover performs well as long as the multi-path distortion is low. Unfortunately, this is often not the case on the lower frequencies -- especially in the summer on short-skip circuits. Thus even with high SNR (>40 dB), Clover is often stuck in its lowest modulation mode for linked circuits (BPSM at 8 to 12 chars/sec depending on the ECC coding rate/efficiency). HAL, unfortunately, has not been able to take advantage of circuits with high SNR to at least partially overcome mutli-path distortion limitations when operating Clover.

Larry  
WA6JYJ/7  
laitinen@eshop.uoregon.edu

---

#### 46. Re: [HFSIG:772] Re: BPSK or DPBSK ?

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 22 Dec 1995 10:50:59 -0800  
To: hfsig@tapr.org  
Message-ID: <9512221848.AA00312@ucs.orst.edu>  
Rendered body type: text/plain

Hi Larry,

I think we had a QSO or two on 2 meters in the past - glad to see you on HFSIG.

```
>
>
>
>>Concerning the discussion on BPSK vs. DBPSK:
>>
>>>
>>> Differential signaling doubles the error rate. As receiver takes the
>>> difference between the previous bit and the current bit, one error in the
>>> channel results in an two errors at the output. In a typical system, at
>>> ^^^^^^
>>> low error rates, this results in a fraction of a dB worse performance.
>> ^^^^^^^^^^^^^^^^^
>>
>>What type of a link is expected for HF? This statement surprised me
>>a bit, because I think of typical HF links as having to work at low SNR
>>(or at least quite variable SNR) and thus high BER.
>
>More importantly, it seems to me, is the multi-path induced phase jitter.
>Many times, particularly on the lower HF freqs, I have experienced links
>with excess SNR (above that which is needed for a low BER) behave poorly
>with Clover due to excessive multi-path distortion.
>
>>
>>I have not been on this list long enough to know what kind of
>>link people expect to work with. My HF experience is limited
>>to CW and RTTY. Signals on RTTY can have good SNR, but you would
>>like a system to work to as low an SNR as possible.
>>
```

>>Gary  
 >>  
 >  
 >Does anyone have experience with HAL's PCI-4000/M in the Clover and  
 >FSK modes? HAL's DSP implementation of FSK filters on the PCI-4000/M  
 >seems to be lacking. In my comparisons with the PK-232, e.g., it  
 >generally performs poorly. The PK-232 is not a particularly good  
 >reference model! My 35 year old vacuum tube TU performs better than  
 >the PK-232 on 45 Baud RATT.

I cannot help here as I have not had the opportunity to play with a PCI-4000/M. I have heard that HAL have made improvements to their RTTY filters to bring their performance on par with the competition. With the hardware on their boards, I don't see any reason why the software cannot be made to equal or better our usual RTTY benchmark modems

However, a few thoughts: Clover uses a four-tone ensemble that is engineered for extremely compact and narrow bandwidth. This narrow bandwidth, however, does not come without penalties and perhaps is the price you have to pay on the long run. 80m, and 40m at times, has extreme cases of multipath. This effect, as we know, plays havoc on absolute phase relationships also includes doppler-effects. The only possible guard against these adverse affects is redundancy - Clover at its lowest data rate makes an attempt to implement redundancy in two ways: 1) by using the four tones redundantly (they all carry the same information), 2) hope that the Reed-Solomon ECC will be effective. At this low rate Clover effectively uses a signaling rate of approximately 31 baud, which, in theory at least, should allow DBPSK to work. It thus remains a question whether the Clover demodulator does a good job of using diversity principles. In other words, does it still use narrow-band techniques for each of the four tones and whether that is a good idea is questionable - perhaps this is what you are alluding to in your comparisons to older tube-type technology. There the situation is a bit different, i.e., much wider bandwidth filters that are most forgiving.

>  
 >Clover performs well as long as the multi-path distortion is low.  
 >Unfortunately, this is often not the case on the lower frequencies --  
 >especially in the summer on short-skip circuits. Thus even with  
 >high SNR (>40 dB), Clover is often stuck in its lowest modulation  
 >mode for linked circuits (BPSM at 8 to 12 chars/sec depending on the  
 >ECC coding rate/efficiency). HAL, unfortunately, has not been able  
 >to take advantage of circuits with high SNR to at least partially  
 >overcome mutli-path distortion limitations when operating Clover.  
 >  
 >Larry  
 >WA6JYJ/7  
 >laitinen@eshop.uoregon.edu  
 >  
 >

Season's Greetings and 73's,

--Johan, KC7WW

P.S. I'll be out of the woods by January 8th and will be participating more actively.

---

## 47. RE: [HFSIG:773] Re: BPSK or DPBSK ?

From: LAITINEN@ESHOP.UOREGON.EDU  
 Date: Fri, 22 Dec 1995 12:42:42 -0800 (PST)  
 To: hfsig@tapr.org  
 Message-ID: <951222124242.d51@ESHOP.UOREGON.EDU>  
 Rendered body type: text/plain  
 Johan:

FB. Yes, I operate two meters and travel up to OSU every couple of months. So we have probably yakked before on one of the repeaters, possibly the Vineyard Mtn repeater...

Yes, HAL improved the DSP FSK filters in the PCI-4000/M. The first release of the /M upgrade was so bad it was essentially unusable. The next release (and I'm going by memory here, so I may not have it quite right) made for usable RATT/Amtor. But performance was poor. I think the Pactor came out in this release and was essentially unusable. The last release that I got brought the RATT mode up to what I originally thought was on par or better than the PK-232. Bugs fixed in the Pactor protocol implementation made it useful! Seems to work real well. But more work with the RATT

mode at my station showed that the PK-232 still outperforms the HAL PCI-4000/M under various conditions. Often times the PK-232 yielded solid copy while the PCI-4000/M was struggling...

So I'm wondering if others have experienced similar results or if I'm the lone ranger???

My comparisons with 1960's technology were between the PK-232 and my vintage W6NRM Mark IV-2 TU with slideback detectors. The Mark IV-2 does a much better job on 45 Baud RATT than does the PK-232. The Mark IV-2 was designed for 45-Baud while the PK-232 has much wider bandwidth... Then there are the ancient slide-back detectors...

As for the Clover mode on the PCI-4000/M: I've had numerous discussions and correspondence with Bill Henry of HAL and Ray Petit about the Clover performance vs SNR and PHS (multipath/phase dispersion). It has been clear from the beginning that Bill wants to go fast under near ideal conditions and does not care much about poor signal conditions. His business considerations (marketing) are the primary reasons for this. Speed, like sex, sells...

My experience also shows that Kantronics' G-TOR often yields about the same net throughput as Clover for similar conditions. Yet the symbol rate for G-TOR is much higher than that of Clover. Although Clover has a much lower symbol rate and in principle should tolerate multi-path better than higher symbol rate systems, this does not always appear to be the case. Perhaps HAL's choice of modulation based on SNR es PHS transition points is too conservative.

Your point about bandwidth is well taken. Bill points to that with considerable pride. Obviously G-TOR using BFSK with the usual implementation has a lousy spectrum -- especially compared to Clover.

So I guess the bottom line for me is that I like Clover very much. I just wish that it would do better under medium (typical 80 meter) multi-path conditions. I hate being stuck in the lowest (BPSM) mode! I have suggested to HAL & Co the exploitation of excess SNR to improve the throughput under high PHS conditions. Bill was not receptive at all.

To recover his development cost he appeared to me (a couple of years ago) to be mainly interested in the FEMA/military/gov't/commercial market for Clover. Where one has point-to-point circuits with rhombics/lpa antennas running 10-KW, etc. HAL es FEMA did some tests between Boulder es WashDC with such equipment and had those puppies running fairly consistently at the 16P4A (highest speed) mode (about 750 bits/sec before the R-5 coding overhead). That is 16 phase shift levels and 4 amplitude shift levels. I believe this was done up around 12-MHz -- a lot different propagation than what I see on 80 meters!

It may be that I am unrealistic in thinking that excess SNR can be used to overcome multi-path problems. Ray did some tests on this a year or so ago and the results were not very good. More work needs to be done on this I think...

Larry  
WA6JYJ/7  
laitinen@eshop.uoregon.edu

---

## 48. Re: [HFSIG:770] Re: BPSK or DBPSK ?

From: Jeffrey Austen <JRA1854@ntech.edu>  
Date: Fri, 22 Dec 1995 17:37:47 -0600 (CST)  
To: hfsig@tapr.org  
Message-ID: <01HZ46K3J1HU9FMUWS@ntech.edu>  
Rendered body type: text/plain

> > Differential signaling doubles the error rate. As receiver takes the  
> > difference between the previous bit and the current bit, one error in the  
> > channel results in an two errors at the output. In a typical system, at  
> > low error rates, this results in a fraction of a dB worse performance.  
> > What type of a link is expected for HF? This statement surprised me  
> > a bit, because I think of typical HF links as having to work at low SNR  
> > (or at least quite variable SNR) and thus high BER.

Yes, at HF the error rate would be higher due mainly to interference and propagation effects. At higher error rates the performance loss due to differential signaling increases. For an example I'll use the equation



$P_e = \exp(-E_b/N_0)$   
 which, although it may not be exact, it is easy to manipulate and is close enough for this purpose. A few results follow:

| $P_e(\text{original})$ | $P_e(\text{differential})$ | $E_b/N_0(\text{orig.})$ | $E_b/N_0(\text{diff.})$ | Difference(dB) |
|------------------------|----------------------------|-------------------------|-------------------------|----------------|
| $10^{-2}$              | $2 \cdot 10^{-2}$          | 4.605                   | 3.912                   | 0.71           |
| $10^{-4}$              | $2 \cdot 10^{-4}$          | 9.210                   | 8.517                   | 0.34           |
| $10^{-6}$              | $2 \cdot 10^{-6}$          | 13.816                  | 13.122                  | 0.22           |

Jeff, k9ja

+--+

Jeffrey Austen | Tennessee Technological University  
 jra1854@tntech.edu | Box 5004  
 (615) 372-3485 | Cookeville Tennessee 38505 U.S.A.

#### 49. Re: [HFSIG:775] Re: BPSK or DBPSK ?

From: Phil Karn <karn@qualcomm.com>  
 Date: Fri, 22 Dec 1995 16:52:27 -0800 (PST)  
 To: hfsig@tapr.org  
 Message-ID: <199512230052.QAA13697@servo.qualcomm.com>  
 Rendered body type: text/plain

I note that on fading channels where it's difficult to track carrier phase, noncoherently demodulated M-ary orthogonal modulation (e.g., M-ary FSK) with reasonably large values of M provide better  $E_b/N_0$  performance than \*any\* binary scheme, including coherently demodulated binary PSK, which in turn is better than noncoherent DPSK.

Phil

#### 50. Re: [HFSIG:768] Re: Negative Frequencies

From: danie.brynard@pixie.co.za (Danie Brynard)  
 Date: Sat, 23 Dec 1995 06:08:03 +0200  
 To: hfsig@tapr.org  
 Message-ID: <199512230408.GAA13291@f15.pix.za>  
 Rendered body type: text/plain

Thank you TOM for explaining it in such nice and clear terms. I have spoken to many people on the Internet about this problem of mine but I think your explanation is one if the better.

I hope other 'dummies' like me has at least also learned something from the discussion :-)

A merry christmas to all.

Thanks again & 73 de danie zs6awk

```
>
>
> Danie: as a follow-up to yesterday's message I broke out the
>freshman physics book, and here's the derivation of the formula:
>
>>Assume two vectors, A and B. Define the vector A (in 3-dimensions)
>as follows:
>
> A = ax X + ay Y + az Z
>
>where X, Y, and Z are the three directions on the coordinate system.
>ax is the component of vector A in the X direction, ay is the component
>in the Y direction, and az is the component in the Z direction.
>We'll assign X to be the real axis, Y to be the imaginary axis, and
>Z to be normal to X and Y (Z points out of the page formed by the real,
>imaginary plane. In the right-handed coordinate system,
>X goes from center to the right, Y goes from the center up the page,
>and Z comes out of the page towards the reader.
>
>>Assuming positive frequency, then vector A (at time t) rotates to
>position B (at time t+1) by making counter-clockwise rotation.
>
>>Then we form the two signals at time (t) and time (t+1) and
>assign each to vector A and B, respectively. Assume that (re1,im1)
>is at time (t) and (re2,im2) is at time (t+1), then:
>
> A = re1 X + im1 Y + 0 Z
> B = re2 X + im2 Y + 0 Z
>
>Note that there is no component of either vector in the Z axis, so
```

```

-----+-----
> Tom McDermott | "All opinions expressed
> Alcatel Network Systems, Inc. | are my own, and do not
> mcdermot@aud.alcatel.com | represent those of Alcatel
> [ICC'96 Technical Program Secretary] | Network Systems, Inc."
> [June 23-27, 1996, Dallas, Tx.] |
> -----+-----
>
>
>

```

## 51. Re: [HFSIG:771] Re: Negative Frequencies

From: danie.brynard@pixie.co.za (Danie Brynard)  
Date: Tue, 26 Dec 1995 14:09:06 +0200  
To: hfsig@tapr.org  
Message-ID: <19512261209.OAA00118@f15.pix.za>  
Rendered body type: text/plain

Thanks Phil. How do you know where a bitinterval starts and ends ? I suppose one get that info from the bitsync if you have implemented one hey ?

73 Danie

```
>Danie,
>
>This method of looking at phasor rotation is probably not particularly
>optimal as a FSK demod. The best FSK demodulator is still a bank of
>correlators where you integrate the energy for each tone over a bit
>interval and then pick the strongest.
>
>Phil
>
>
```

## 52. More Pentium DSP Benchmarks

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Tue, 26 Dec 1995 17:03:39 -0800  
To: HFSIG@tapr.org  
Message-ID: <9512270100.AA30611@ucs.orst.edu>  
Rendered body type: text/plain

Hi,

I recently received my IAL (Intel Architecture Labs) CDROM, mainly to look at Intel's NSP (native signal processing). For those not familiar with this - it's an attempt to use the main cpu for doing math intensive tasks usually associated with DSP processors. NSP version 2.0 is included on the CDROM - supplied as library code modules for either Microsoft or Borland compiler tools. A comprehensive on-line reference manual and a few DSP applications are also included.

This library may be of interest to amateur radio DSP experimentors and will run on a i386 or higher processor, however, a Pentium will be best for several reasons (it appears to work just fine on my AMD DX4-120, but I have not done any extensive testing). The following benchmarks are shown for a 90 MHz Pentium to illustrate its capabilities:

| Function                                | Time    |
|-----------------------------------------|---------|
| 256 point Complex FFT                   | 256 uS  |
| 256 point REal FFT                      | 184 uS  |
| 1024 point Complex FFT                  | 1413 uS |
| 1024 pint Real FFT                      | 831 us  |
| 512 point Complex Vector multiplication | 59 uS   |
| 1024 point Complex dot product          | 321 uS  |
| 1024 point Real dot prod.               | 32 uS   |
| 32 tap Complex FIR                      | 5.03 uS |
| 32 tap Real FIR                         | 1.57 uS |

(uS=microseconds)  
(Complex=single or double precision floating point numbers)  
(Time computed by repeated calling of the function, i.e., takes into account parameter passing overheads etc.)

These performance figures shows that modest DSP applications can be performed on such a Pentium processor. The included examples on the CDROM shows fairly impressive real-time performance under the Win95/Win NT OS's. Unfortunately, no library source code is supplied. I found it a simple matter to build applications using the supplied libraries using the Microsoft 32-bit compiler.

Anyone interested in obtaining the CD-ROM may try call Intel at 1-800-253-3696, select option "1" and ask for product code Q495CD (US or Canada only) or 1-503-264-2203 (outside US). This is the IAL developer's network - it's free and comes with a demo version of SPRY's Web browser and

Adobe Acrobat for browsing the CD's contents.

Hope this is of interest for those brave explorers on HFSIG.

--Johan Forrer, KC7WW

---

### 53. Re: [HFSIG:717] BPSK modem for satellites

From: "David M. Gaytko / WD4KPD" <cn2859@abaco.coastalnet.com>  
Date: Wed, 27 Dec 95 19:33:19 EST  
To: hfsig@tapr.org  
Message-ID: <9512280033.AB12015@abaco.coastalnet.com>  
Rendered body type: text/plain

At 05:05 PM 11/27/95 -0600, you wrote:

>Hi all,

>

>I have made up a 400 bps BPSK modem (for A0-13 telemetry) and I plan to  
>scale it to the 1200 bps version. Is there anybody interrested out there  
>who could verify its operation and compare it against some other designs ?  
>The modem runs on the EVM56K or on the DSPCARD4.

>I will try to make tests with my radio equipment but my only VHF/UHF  
>antenna is omnidirectional thus the satellite signals are rather weak.

>

>Pawel

>=====

>=====

==hello, been following your messages about ur psk modem. am a paket

enthusiast, and want use the psk modem on meteor scatter. would like to hear more about ur system, and if it may be available?

tk& 73's....david

>  
>

---

## 54. (no subject)

From: (unknown sender)  
Date: (date unavailable)  
Rendered body type: text/plain

From: JALOCHA@chopin.ifj.edu.pl  
Received: from CHOPIN.IFJ.EDU.PL (chopin.ifj.edu.pl [192.86.14.9]) by nms.cyf-kr.edu.pl (8.6.11/8.6.11) with SMTP id MAA11091 for <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>; Thu, 28 Dec 1995 12:10:19 +0100  
Date: Thu, 28 Dec 1995 12:06 GMT+1  
Subject: Re: [HFSIG:780] Re: BPSK modem for satellites  
To: hfsig <@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>  
Message-id: <A5570B3FC02218DC@chopin.ifj.edu.pl>  
X-Envelope-to: @NMS.CYF-KR.EDU.PL:hfsig@tapr.org  
X-VMS-To: IN% "<@NMS.CYF-KR.EDU.PL:hfsig@tapr.org>"

Hello David,

>==hello, been following your messages about ur psk modem. am a paket  
>enthusiast, and want use the psk modem on meteor scatter. would like to hear  
>more about ur system, and if it may be available?

The most recent source you can find at wooster.hut.fi in pub/alefnul/incoming (an FTP site for quick exchange of the DSPCARD4 software).  
The BPSK1200.ASM is a 1200 bps BPSK modem. Its parameters are preset for satellite work but they can be easily modified for simplex mode.

The other file SLOWBPSK.ASM is the 30 bps BPSK modem with ASCII terminal interface.

I don't know which speed you need for meteor scatter ?  
Do meteor tails keep reasonable phase coherency ?  
If not, the FSK with non-coherent demodulator would be a better solution.

Pawel

---

## 55. Re: [HFSIG:778] Re: Negative Frequencies

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 28 Dec 1995 20:00:12 -0800 (PST)  
To: hfsig@tapr.org  
Message-ID: <199512290400.UAA08846@servo.qualcomm.com>  
Rendered body type: text/plain

>Thanks Phil. How do you know where a bitinterval starts and ends ? I suppose  
>one get that info from the bitsync if you have implemented one hey ?

Yeah. Acquiring bit timing is probably exceeded in difficulty only by acquiring carrier phase. But while you can punt on the latter by using a noncoherent demodulation technique, you really can't avoid the former.

There are many ways discussed in the literature, and the principles are somewhat similar. The job is made somewhat easier than carrier phase acquisition by the much lower frequency of the data rate (as opposed to the RF carrier frequency) and the correspondingly smaller uncertainty due to doppler shift and other distortions. At extremely low data rates you can use a common external reference (e.g., GPS) but only when the propagation delay is small or easily characterized.

Phil

---

## 56. Re: [HFSIG:779] More Pentium DSP Benchmarks

From: Phil Karn <karn@qualcomm.com>  
Date: Thu, 28 Dec 1995 20:42:39 -0800 (PST)  
To: hfsig@tapr.org  
Message-ID: <199512290442.UAA08925@servo.qualcomm.com>  
Rendered body type: text/plain

Johan,

This package sounds very interesting. And did I read you correctly that the price is right, i.e., free?

>256 point Complex FFT        256 uS

This seems too good to be true. Sure it's a P90?

I sweat blood to get my own floating point 256-point CFFT down to 335 microseconds on the P90, and that involved doing a radix-4 primitive in assembler that I groveled over for days (rewriting it several times) to keep the FPU pipeline full. And that's without bit reversal, which I kept separate because it can often be avoided in practice (e.g., in fast convolution).

>32 tap Real FIR                1.57 uS

This seems more realistic. I did a dot product (FIR) that executes in 3 clocks per tap once the pipe fills. This is about as fast as I think it's possible to go on the Pentium given its instruction set -- unless there's a really clever trick I haven't discovered yet.

Phil

---

## 57. Radio Mail using PACKTOR/GTOR

From: FLundgren@daniel.maf.org (Fred Lundgren)  
Date: Fri, 29 Dec 95 06:55:43 UTC  
To: hfsig@tapr.org  
Message-ID: <9511298202.AA820245592@gabriel.maf.org>  
Rendered body type: text/plain

This is a request for help on the subject of interfacing TOR mode communications in the 4 to 7Mhz region with Lotus CCMail, which in turn uses telephone modems to communicate over standard telephone lines to our headquarters in Redlands CA.

First by of introduction: My name is Fred Lundgren(N7LSZ, ER0FL, EK9FL). I am a missionary with the Information Technology(IT) branch of Mission Aviation Fellowship(MAF), based in Redlands CA. MAF's primary mission is to provide air service to remote parts of the world. This amounts to flying missionaries, medical supplies and food, etc. to remote parts on the world. An example of this was demonstrated during the latest conflict between Zaire and Rwanda where MAF flew hundreds of hours carrying food, medical personal and other supplies to the refugee camps in Rwanda.

About a year ago MAF decided that in addition to flight services, remote locations of the world also needed communications. Many of the locations where missionaries live and work have no telephones or electricity. In Zaire for example, MAF establish a central post office or HUB in Kinshasa where land lines were available. Using Lotus CCMail as the E-Mail software and VHF packet using standard AX.25 protocol, they were able to link various remote sites around the city to the home office in Redlands CA. The system knows as MAFnet worked both ways with the Kinshasa post office HUB being transparent to the operators at either end.

This system worked quite well locally. However, what happens when we try to connect Nyenkundi 350 miles to the north east to the Kinshasa post office HUB. HF seemed the obvious solution, unless of course we wanted to install several repeaters in between. Unlike VHF/AX.25, the protocol for HF is not so easy. At this point I am embarrassed to say I'm in the dark as to exactly why G-Tor for instance does not interface directly to CCMail the way VHF packet does. Our main techie has spent hundreds of hours writing code to accomplish this interface. His efforts, for the most, part have been successful however, the process is getting very complicated. This is the one thing we try not to do in remote parts on the world where the operators are non-technical. It's hard to believe there is not a commercial product or technology out there that will interface HF packet directly to CCMail without all the fuss in-between. Others have recommended TCP/IP, JNOS, LINUX, dual port OS/2 etc. however when it comes down to doing it, the flow of information stops.

Therefor, after all that palaver, my question is, and I'm not sure I can even asked it intelligently:

If you were to type out a message on your computer and then sent the message over HF packet radio, which in turn would be interfaced to the telephone at the post office HUB, what HF

---

## 58. Re: [HFSIG:779] More Pentium DSP Benchmarks

From: gc@fox.cen.com (Gary Chatters)  
Date: Fri, 29 Dec 95 10:10:31 EST  
To: hfsig@tapr.org  
Message-ID: <9512291510.AA03772@cen.com>  
Rendered body type: text/plain

>  
>I recently received my IAL (Intel Architecture Labs) CDRom, mainly to look  
>at Intel's NSP (native signal processing). For those not familiar with this

IAL does have web pages. The home page is at <http://www.intel.com/ial>.

They publish quarterly CD-ROM and an on-line newsletter. The May 1995 issue discusses the NSP library (at least, they say it does. I get an error when I try to display it, but I can display other issues). It does require Adobe Acroreader.

There is an on-line order form for getting the latest CD-ROM. I have requested a copy; I'll find out later if it is the right one. Maybe it would be easier to just call. :-)  
They have a lot of free stuff and other things they charge for. I tend to think of Intel as "hardware", but they do have some interesting and useful software, too.

73,

Gary

---

## 59. Re: [HFSIG:783] Re: More Pentium DSP Benchmarks

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 29 Dec 1995 12:46:00 -0800  
To: hfsig@tapr.org  
Message-ID: <9512292042.AA29476@ucs.orst.edu>  
Rendered body type: text/plain

Hi Phil,

>Johan,  
>  
>This package sounds very interesting. And did I read you correctly that  
>the price is right, i.e., free?

I filled out the form on Intel's Web page and the CDRom arrived a few days later by UPS - really excellent customer service. Free!

>  
>>256 point Complex FFT        256 uS  
>  
>This seems too good to be true. Sure it's a P90?  
>

These performance figures are presented in a document on the CD in the directory NSP20\DOC. It says:

"All results averaged over large repetitions of the function call. Data in cache. Results represent an upper bound as some timing and operating system overhead is included. 90 MHz Pentium with 256K 2nd level cache under Win NT 3.50"

Of course, paper is patient and perhaps you could pose the question - I get the sense that they are keen to please and invite feedback.

>I sweat blood to get my own floating point 256-point CFFT down to 335  
>microseconds on the P90, and that involved doing a radix-4 primitive  
>in assembler that I groveled over for days (rewriting it several  
>times) to keep the FPU pipeline full. And that's without bit reversal,  
>which I kept separate because it can often be avoided in practice  
>(e.g., in fast convolution).  
>  
>>32 tap Real FIR                1.57 uS  
>  
>This seems more realistic. I did a dot product (FIR) that executes in  
>3 clocks per tap once the pipe fills. This is about as fast as I think  
>it's possible to go on the Pentium given its instruction set -- unless

>there's a really clever trick I haven't discovered yet.  
>  
>Phil  
>  
>

The library is rich in functionality for making all sorts of trig conversions: bit reversing and other twiddle factoring issues used in FFT's, Goertzel's algorithm, windowing functions, multirate processing, and LMS. Since you have been playing with similar code, it may be worth looking into.

Good luck,

--Johan

---

## 60. Re: [HFSIG:784] Radio Mail using PACKTOR/GTOR

From: forrerj@ucs.orst.edu (Johan Forrer)  
Date: Fri, 29 Dec 1995 12:46:07 -0800  
To: hfsig@tapr.org  
Message-ID: <9512292042.AA29494@ucs.orst.edu>  
Rendered body type: text/plain

Dear Fred,

> This is a request for help on the subject of interfacing TOR mode  
> communications in the 4 to 7Mhz region with Lotus CCMail, which in  
> turn uses telephone modems to communicate over standard telephone  
> lines to our headquarters in Redlands CA.

Welcome to HFSIG.

>  
> First by of introduction: My name is Fred Lundgren(N7LSZ, ER0FL,  
> EK9FL). I am a missionary with the Information Technology(IT) branch  
> of Mission Aviation Fellowship(MAF), based in Redlands CA. MAF's  
> primary mission is to provide air service to remote parts of the  
> world. This amounts to flying missionaries, medical supplies and  
> food, etc. to remote parts on the world. An example of this was  
> demonstrated during the latest conflict between Zaire and Rwanda where  
> MAF flew hundreds of hours carrying food, medical personal and other  
> supplies to the refugee camps in Rwanda.  
>

> About a year ago MAF decided that in addition to flight services,  
> remote locations of the world also needed communications. Many of the  
> locations where missionaries live and work have no telephones or  
> electricity. In Zaire for example, MAF establish a central post  
> office or HUB in Kinshasa where land lines were available. Using  
> Lotus CCMail as the E-Mail software and VHF packet using standard  
> AX.25 protocol, they were able to link various remote sites around the  
> city to the home office in Redlands CA. The system knows as MAFnet  
> worked both ways with the Kinshasa post office HUB being transparent  
> to the operators at either end.  
>

> This system worked quite well locally. However, what happens when we  
> try to connect Nyenkundi 350 miles to the north east to the Kinshasa  
> post office HUB. HF seemed the obvious solution, unless of course we  
> wanted to install several repeaters in between. Unlike VHF/AX.25, the  
> protocol for HF is not so easy. At this point I am embarrassed to say  
> I'm in the dark as to exactly why G-Tor for instance does not interface  
> directly to CCMail the way VHF packet does. Our main techie has spent  
> hundreds of hours writing code to accomplish this interface. His  
> efforts, for the most, part have been successful however, the process  
> is getting very complicated. This is the one thing we try not to do in  
> remote parts on the world where the operators are non-technical. It's  
> hard to believe there is not a commercial product or technology out  
> there that will interface HF packet directly to CCMail without all the  
> fuss in-between. Others have recommended TCP/IP, JNOS, LINUX, dual  
> port OS/2 etc. however when it comes down to doing it, the flow of  
> information stops.  
>

> Therefor, after all that palaver, my question is, and I'm not sure I  
> can even asked it intelligently:

>  
> If you were to type out a message on your computer and then  
> sent the message over HF packet radio, which in turn would be  
> interfaced to the telephone at the post office HUB, what HF  
> protocol would you need?  
>  
>

Just my 2 cent's worth.

HF TOR protocols, defines a physical transport level where the TOR protocol logic (AMTOR, PACTOR, GTOR etc.) is typically implemented on dedicated microcontrollers that is part of TNC hardware. This is quite common for example the PK-232's, KAM, etc. It appears there is no standard as far as TNC to host protocol is concerned, i.e., everyone seems to have their own proprietary "host mode" which is your main problem. For VHF networking, "KISS" protocol is pretty much universally adopted as a TNC-host protocol. This means that you will have an easier task in setting up networking client/server systems on VHF. To obtain similar networking implementations on HF with the status quo, you have a problem - they don't handle KISS on HF TOR - the overhead could be render it useless.

Now, if you insist on going this route using off-the-shelf HF-capable TNC's, there are two ways to handle the networking interfacing issue - the solution depends on whether you are going to deal with a single manufacturer's TNC or whether you need to be able to use a number of different manufacturer's TNC's. One solution is to write the necessary driver software on the host to buffer and translate between the networking side and the HF TNC side. The other is to develop a "black box" that has the ability to handle a KISS port on the one end and provide a TNC-specific port on the other. Even a modest microcontroller would be sufficient in the latter situation. Neither solution, however, is trivial and will need careful consideration and probably cost you real money.

You may be interested in the parallel HF modems that have been discussed over the past year on this list. In the present form, the author, Pawel Jolocha (SP9VRC), designed these to use a common KISS interface to the host and thus makes the server side much simpler. These modems use AX.25 for the physical transport layer. This kind of approach probably will get you going really fast in the short term, however, we have yet to discuss in all seriousness, the issue of a protocol for the physical transport layer for HF.

I can think of further solutions - however let see how this goes for starters.

--Johan, KC7WW

---

## 61. Re: [HFSIG:785] Re: More Pentium DSP Benchmarks

From: gc@fox.cen.com (Gary Chatters)  
Date: Fri, 29 Dec 95 15:43:34 EST  
To: hfsig@tapr.org  
Message-ID: <9512292043.AA10064@cen.com>  
Rendered body type: text/plain

I wrote:

>  
>IAL does have web pages. The home page is at <http://www.intel.com/ial>.  
>  
>They publish quarterly CD-ROM and an on-line newsletter. The May 1995  
>issue discusses the NSP library (at least, they say it does. I get an  
>error when I try to display it, but I can display other issues).

The problem was an old version of Acrobat Reader. The current version (2.1) works fine with the May 1995 issue. And it does have a couple of articles on NSP.

Gary

---

## 62. Nordic HF Conference

From: lay@cod.nosc.mil (Richard Lay)  
Date: Fri, 29 Dec 95 14:58:53 PST  
To: hfsig@tapr.org  
Message-ID: <9512292258.AA24448@marlin.nosc.mil>  
Rendered body type: text/plain

Hi all!

Has anyone checked out the Nordic HF Conference web pages at <http://147.13.228.1/Radio/NordicHF/> ? If so, has anyone successfully downloaded, unzipped, and viewed any of the conference papers (they're zipped, postscript files)? In particular, "Description of a Laboratory Jammer for Automatic HF Systems" by P Nilsson, Proceedings of the Nordic Shortwave Conference HF95, 15-17 August 1995. Editor Olov Carlsson. Published by the Nordic Radio Society. The first five pages look interesting. (That's all I've been able to view. Anyone else have better luck?)



Happy Holidays!

Rich

---

### 63. Re: [HFSIG:789] Nordic HF Conference

From: chbrain@dircon.co.uk (Charles Brain)  
Date: Sat, 30 Dec 1995 10:29:00 +0000  
To: hfsig@tapr.org  
Message-ID: <199512301050.AA00467@felix.dircon.co.uk>  
Rendered body type: text/plain

>Hi all!  
>  
>Has anyone checked out the Nordic HF Conference web pages at  
><http://147.13.228.1/Radio/NordicHF/> ? If so, has anyone successfully  
>downloaded, unzipped, and viewed any of the conference papers (they're  
>zipped, postscript files)? In particular, "Description of a Laboratory  
>Jammer for Automatic HF Systems" by P Nilsson, Proceedings of the Nordic  
>Shortwave Conference HF95, 15-17 August 1995. Editor Olov Carlsson.  
>Published by the Nordic Radio Society. The first five pages look  
>interesting. (That's all I've been able to view. Anyone else have better  
>luck?)  
>  
>Happy Holidays!  
>  
>Rich  
>  
>  
Hello Rich,  
No I haven't tried it, but we have a copy of all the proceedings at work as  
the company I work for presented three papers one on H.F link  
management another on managed networks (use of SNMP to control  
large H.F Radio Systems) and also I think a paper on our Digital H.F  
receiver and drive. If you are having trouble try Emailing Hakan Bergzen at  
[Hakan.Bergzen@telub.se](mailto:Hakan.Bergzen@telub.se) he watches this news group so he may reply  
directly.

Regards Charles

---